28

Access DB# 116435

SEARCH REQUEST FORM

Scientific and Technical Information Center

Requester's Full Name: Michael Lewis Examiner # : 80141 Date: 3/9/04
Art Unit: 2655 Phone Number 30 Serial Number: 09 816 032
Mail Box and Bldg/Room Location: ~~PK 2~~ 8D50 Results Format Preferred (circle): PAPER DISK E-MAIL

If more than one search is submitted, please prioritize searches in order of need.

****

Please provide a detailed statement of the search topic, and describe as specifically as possible the subject matter to be searched. Include the elected species or structures, keywords, synonyms, acronyms, and registry numbers, and combine with the concept or utility of the invention. Define any terms that may have a special meaning. Give examples or relevant citations, authors, etc, if known. Please attach a copy of the cover sheet, pertinent claims, and abstract.

Title of Invention: Voice Encoding Apparatus & method thereof

Inventors (please provide full names): Fumio Amano

Earliest Priority Filing Date: 3/22/01

**For Sequence Searches Only* Please include all pertinent information (parent, child, divisional, or issued patent numbers) along with the appropriate serial number.*

See claim one 1

3-9-04 2:15pm

****

STAFF USE ONLY	**Type of Search**	**Vendors and cost where applicable**
Searcher: Pamela Reynolds	NA Sequence (#)	STN
Searcher Phone #: 306-0255	AA Sequence (#)	Dialog ✓
Searcher Location: PK2 3C03	Structure (#)	Questel/Orbit
Date Searcher Picked Up: 3-16-04 10W	Bibliographic ✓	Dr.Link
Date Completed: 3-16-04 3N	Litigation	Lexis/Nexis
Searcher Prep & Review Time: 82	Fulltext	Sequence Systems
Clerical Prep Time:	Patent Family	WWW/Internet
Online Time: 128	Other	Other (specify) ✓

PTO-1590 (8-01)

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File 344:Chinese Patents Abs Aug 1985-2004/Mar
        (c) 2004 European Patent Office
File 347:JAPIO Nov 1976-2003/Nov(Updated 040308)
        (c) 2004 JPO & JAPIO
File 348:EUROPEAN PATENTS 1978-2004/Mar W01
        (c) 2004 European Patent Office
File 349:PCT FULLTEXT 1979-2002/UB=20040311,UT=20040304
        (c) 2004 WIPO/Univentio
File 350:Derwent WPIX 1963-2004/UD,UM &UP=200417
        (c) 2004  Thomson Derwent
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Set	Items	Description
S1	232	AU=(AMANO, F? OR AMANO F?)
S2	1	S1 AND VOICE()DECOD?
S3	10	S1 AND VOICE()ENCOD?
S4	10	S3 NOT S2
S5	4	S4 AND FRAME?
S6	4	S5 NOT S2

2/5/1 (Item 1 from file: 347)
DIALOG(R)File 347:JAPIO

03784500 **Image available**
VOICE DECODING SYSTEM

PUB. NO.: 04-149600 [JP 4149600 A]
PUBLISHED: May 22, 1992 (19920522)
INVENTOR(s): TANAKA YOSHIAKI
TANIGUCHI TOMOHIKO
OTA TAKASHI
AMANO FUMIO
APPLICANT(s): FUJITSU LTD [000522] (A Japanese Company or Corporation), JP (Japan)
APPL. NO.: 02-274834 [JP 90274834]
FILED: October 12, 1990 (19901012)
INTL CLASS: [5] G10L-009/00; G10L-009/14; G10L-009/18
JAPIO CLASS: 42.5 (ELECTRONICS -- Equipment)
JAPIO KEYWORD:R108 (INFORMATION PROCESSING -- Speech Recognition & Synthesis)
JOURNAL: Section: P, Section No. 1419, Vol. 16, No. 436, Pg. 5, September 11, 1992 (19920911)

ABSTRACT

PURPOSE: To detect an error in pitch cycle on a reception side without adding redundancy to sent data by delaying the data on a decoding side by a pitch period D received from an encoding side.

CONSTITUTION: On the encoding side, the maximum pitch period D among pitch periods in a pitch period search range and the gain (g) at the maximum pitch period are found, and multiplexed with other parameters and sent to a decoder. The pitch period D is received by the decoder and supplied to a buffer 40. Then pitch vectors P(sub 1) and P(sub 2) of the pitch period D and a pitch period 2D which is double as long as it are extracted from an adaptive code book 10 and the normalization correlation coefficient r(sub 12) between the both is calculated from an equation V(sub 12)=(P(sub 1),P(sub 2))/(P(sub 1),P(sub 1)). Then an error presence/absence detection part 50 receives the normalization correlation coefficient r(sub 12) from the buffer 40 to detect whether there is an error or not. The correlation coefficient r(sub 12) is compared with with its threshold value r(sub th) and when r(sub 12)<r(sub th), it is judged that the period D has an error, so that a repair processing part 60 performs repair processing.
?

6/5/1 (Item 1 from file: 347)
DIALOG(R)File 347:JAPIO

07294524 **Image available**
VOICE ENCODING METHOD ACCOMPANIED BY PACKET REPAIR PROCESSING

PUB. NO.: 2002-162998 [JP 2002162998 A]
PUBLISHED: June 07, 2002 (20020607)
INVENTOR(s): **AMANO FUMIO**
APPLICANT(s): FUJITSU LTD
APPL. NO.: 2000-361874 [JP 2000361874]
FILED: November 28, 2000 (20001128)
INTL CLASS: G10L-019/00; G10L-019/12; G10L-019/04; H03M-007/30

ABSTRACT

PROBLEM TO BE SOLVED: To provide a **voice encoding** method accompanied by such packet repair processing that S/N and subjective quality are good and the voice in a consonant section is articulate.

SOLUTION: Multiple interpolation repair processes are prepared on a transmission side. Assuming that **frames** to be transmitted are lost on the transmission side, all the interpolation repair processes are tried, **frame** by **frame** . Then the waveform interpolated and repaired through the repair processes is compared with a reproduced waveform locally decoded from the packet. Then the index number of an interpolation repair processing system which can obtain an interpolated and repaired waveform closest to the locally decoded reproduced waveform is sent to a reception side together with the packet. On the reception side, multiple interpolation repair processes are prepared as well as the transmission side and if the lost of a packet is detected, a interpolation repair system is selected according to the index number of the interpolation repair system sent together with the **frame** and the interpolation repair process is carried out. Consequently, when no packet is lost, the interpolated and repaired waveform which is closest to the decoded reproduced waveform is obtained.

6/5/2 (Item 2 from file: 347)
DIALOG(R)File 347:JAPIO

03912700 **Image available**
VOICE ENCODING SYSTEM

PUB. NO.: 04-277800 [JP 4277800 A]
PUBLISHED: October 02, 1992 (19921002)
INVENTOR(s): OTA TAKASHI
TANIGUCHI TOMOHIKO
TANAKA YOSHIAKI
KURIHARA HIDEAKI
AMANO FUMIO
APPLICANT(s): FUJITSU LTD [000522] (A Japanese Company or Corporation), JP (Japan)
APPL. NO.: 03-040082 [JP 9140082]
FILED: March 06, 1991 (19910306)
INTL CLASS: [5] G10L-009/18; G10L-009/14
JAPIO CLASS: 42.5 (ELECTRONICS -- Equipment)
JAPIO KEYWORD:R108 (INFORMATION PROCESSING -- Speech Recognition &

Synthesis)
JOURNAL: Section: P, Section No. 1486, Vol. 17, No. 72, Pg. 62, February 12, 1993 (19930212)

ABSTRACT

PURPOSE: To enable real-time processing by reducing the arithmetic quantity.

CONSTITUTION: The **voice encoding** system identifies the target vector of an input voice with the vector generated by processing a signal vector, generated by delaying the residue signal of a precedent **frame** , by linear predictive weighted filtering and the vector generated by processing a residue signal read out of a code book (70) by linear predictive weighted filtering. The same sample position of all residue signal vectors in the code book (70) are forcibly set to zero. When the residue vectors are weighted and filtered, the arithmetic for the sample positions of the zero value of the residue signal vectors is omitted.

6/5/3 (Item 3 from file: 347)
DIALOG(R)File 347:JAPIO

03122732 **Image available**
TIME SWITCHING TYPE BAND SPLIT **VOICE ENCODER**

PUB. NO.: 02-098232 [JP 2098232 A]
PUBLISHED: April 10, 1990 (19900410)
INVENTOR(s): **AMANO FUMIO**
OTA TAKASHI
UMIGAMI SHIGEYUKI
APPLICANT(s): FUJITSU LTD [000522] (A Japanese Company or Corporation), JP (Japan)
APPL. NO.: 63-251242 [JP 88251242]
FILED: October 05, 1988 (19881005)
INTL CLASS: [5] H04B-014/04; H04J-003/02
JAPIO CLASS: 44.2 (COMMUNICATION -- Transmission Systems)
JOURNAL: Section: E, Section No. 947, Vol. 14, No. 308, Pg. 28, July 03, 1990 (19900703)

ABSTRACT

PURPOSE: To prevent deterioration in the quality of a reproduced voice signal without increasing much of quantized noise by providing M-set of multiplexer section, arranging outputs of N-set of coders to M-set of multiplex sections sequentially, multiplexing them and outputting outputs of the M-set of multiplex sections while being switching by a switching control section in the unit of **frames** .

CONSTITUTION: A band of an input voice signal is split by N-set of band split filters 1-1-1-N, the split N-set of outputs are interleaved by sample interleave processing sections 2-1-2-N respectively, the interleaved outputs are coded by coders 3-1-3-N and sent with multiplexing. Then the M-set of multiplex sections 4-1-4-N are provided and the outputs of the N-set of coders are arranged and multiplexed in the M-set of the multiplex sections 4-1-4-N and the outputs of the M-set of the multiplex sections are outputted with switching in a **frame** unit. Thus, quantized noise is not so much increased and the quality of the reproduced voice signal is not so much deteriorated.

6/5/4 (Item 1 from file: 350)
DIALOG(R)File 350:Derwent WPIX

014777437 **Image available**
WPI Acc No: 2002-598143/200264
XRPX Acc No: N02-474312

Voice encoding **method for voice over IP network, involves calculating signal-to-noise ratio for each interpolated** frame **equivalent to initial** frame **, to define index number indicating maximum signal-to-noise ratio**

Patent Assignee: FUJITSU LTD (FUIT); AMANO F (AMAN-I)
Inventor: **AMANO F**
Number of Countries: 002 Number of Patents: 002
Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
US 20020065648	A1	20020530	US 2001816032	A	20010322	200264	B
JP 2002162998	A	20020607	JP 2000361874	A	20001128	200264	

Priority Applications (No Type Date): JP 2000361874 A 20001128
Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
US 20020065648	A1		18	G10L-019/00	
JP 2002162998	A		11	G10L-019/00	

Abstract (Basic): US 20020065648 A1

NOVELTY - A **frame** having voice data is encoded and interpolated repeatedly. The signal-to-noise ratio for each **frame** is calculated, based on the interpolated **frames** equivalent to initial **frame** . The index number indicating highest signal-to-noise ratio is calculated and multiplexed with encoded parameters.

DETAILED DESCRIPTION - An INDEPENDENT CLAIM is included for **voice encoder** .

USE - For voice over internet protocol (VOIP) network.

ADVANTAGE - Ensures obtaining high voice quality due to provision of high signal-to-noise ratio.

DESCRIPTION OF DRAWING(S) - The figure shows the block diagram of voice transmission system.

pp; 18 DwgNo 5A/15

Title Terms: VOICE; ENCODE; METHOD; VOICE; IP; NETWORK; CALCULATE; SIGNAL; NOISE; RATIO; INTERPOLATION; **FRAME** ; EQUIVALENT; INITIAL; **FRAME** ; DEFINE; INDEX; NUMBER; INDICATE; MAXIMUM; SIGNAL; NOISE; RATIO
Derwent Class: P86; W01; W04
International Patent Class (Main): G10L-019/00
International Patent Class (Additional): G10L-019/04; G10L-019/12; H03M-007/30
File Segment: EPI; EngPI
?

File 256:SoftBase:Reviews,Companies&Prods. 82-2004/Feb

Set	Items	Description
S1	8135	VOICE OR AUDIO OR SOUND OR SPEECH
S2	4397	FRAME?
S3	14	(INTERPOLAT? OR ENCOD?) AND (REPEAT? OR ITERATIV? OR REDUNDANT? OR REITERA?)
S4	0	S3 AND (RECOVER? OR ERROR?)
S5	5	SIGNAL(3N)NOISE()RATIO
S6	1158	VOIP OR VOICE(3N)INTERNET
S7	7	S2 AND (FIRST OR INITIAL?) AND (SECOND OR SUBSEQUENT?) AND ENCOD?
S8	40	PARAMETER? AND PACKET?
S9	39	CONSONANT??
S10	31	(INDEX OR SEQUENCE)(3N)NUMBER??
S11	0	S10 AND (MULTIPLEX? OR MULTI()PLEX?) AND (TRANSMIT? OR TRANSMIS? OR SEND OR SENDING OR SENDS)
S12	315	(SAME OR CLOSE OR EQUAL OR EQUIVALENT OR APPROXIMAT? OR MAXIMUM OR HIGHEST) AND MATCH?
S13	0	AU=(AMANO, F? OR AMANO F ?)
S14	0	S8 AND S9 AND S10 AND S12
S15	0	(S1 OR S6) AND S8 AND S9 AND S10
S16	9	(S1 OR S6) AND (S8 OR S9 OR S10)
S17	0	S16 AND (S3 OR S7 OR S12)
S18	1	(S1 OR S2) AND S5
S19	0	S8 AND S9 AND S10

16/3,K/1
DIALOG(R)File 256:SoftBase:Reviews,Companies&Prods.

00143686 DOCUMENT TYPE: Review

PRODUCT NAMES: Acterna DA-3400 (148911); Brix Networks Verifiers (148938); Vivinet Manager 2.1 (147583)

TITLE: Sizing up VoIP **listening tools:** VoIP **traffic analysis products...**
AUTHOR: Mier, Edwin
SOURCE: Network World, v19 n49 p47(3) Dec 9, 2002
ISSN: 0887-7661
HOMEPAGE: http://www.nwfusion.com

RECORD TYPE: Review
REVIEW TYPE: Product Comparison
GRADE: Product Comparison, No Rating

REVISION DATE: 20030330

TITLE: Sizing up VoIP **listening tools:** VoIP **traffic analysis products......**

...NetIQ's Vivinet Manager 2.1 are among compared products highlighted in this discussion of **Voice** -over-IP (**VoIP**) traffic analyzers. No vendor yet provides a full set of products that supports all the following: real-time **VoIP** traffic monitoring and alarm generation; long-term **VoIP** activity recording and reporting; **VoIP** traffic generation; **VoIP** node availability; and **VoIP** bandwidth assessment; automated **voice** -quality assessment; measurement of QoS (quality of service) **parameters** ; **VoIP** traffic and protocol decode; and intelligent diagnosis of service problems. NetIQ and Brix provide excellent **VoIP** quality assessment and QoS measurement, while Sniffer Technologies provides the best **VoIP** traffic and protocol decode. Acterna DA-3400 with EtherNet Analysis Software 1.1 and optional **VoIP** Analysis software is best suited for real-time **VoIP** monitoring and QoS assessment. It provides detailed views of **VoIP** activity that emphasize throughput, jitter, and **packet** loss. Finistar Surveyor 5.0. with the THGs monitor/analyzer with Multi-QoS software (which is one of the best real-time **VoIP** monitor software packages reviewed an integrates with Surveyor 5.0) is best suited for real-time monitoring and alarm generation, as well as **VoIP** traffic and protocol decode.

DESCRIPTORS: Internet Traffic Analysis; Network Administration; Network Management; Network Software; Performance Monitors; System Monitoring; **VoIP**

16/3,K/2
DIALOG(R)File 256:SoftBase:Reviews,Companies&Prods.

00142829 DOCUMENT TYPE: Review

PRODUCT NAMES: Adobe Premiere 6.5 (350591)

TITLE: Adobe Ponies Up Real-Time Effects with Premiere 6.5
AUTHOR: Peters, Oliver
SOURCE: Videography, v27 n10 p76(2) Oct 2002

ISSN: 0363-1001
HOMEPAGE: http://www.videography.com

RECORD TYPE: Review
REVIEW TYPE: Review
GRADE: A

REVISION DATE: 20030330

...for the Macintosh and PC platforms, gets excellent scores. The graphical user interface (GUI) is **consonant** with other Adobe products', but could use some freshening and a more OS X or...

...for source clips. Users can stack as many as 99 video tracks and 99 stereo **audio** tracks. A project and a sequence are identical; multiple sequences cannot be in the same...

16/3,K/3
DIALOG(R)File 256:SoftBase:Reviews,Companies&Prods.

00142447 DOCUMENT TYPE: Review

PRODUCT NAMES: NetHawk 5.0 (028185); HYDRA (140384)

TITLE: Security products aim to make nets hacker-proof
AUTHOR: Murray, Charles J
SOURCE: Electronic Engineering Times, v1234 p6(1) Sep 2, 2002
ISSN: 0192-1541
HOMEPAGE: http://www.eet.com

RECORD TYPE: Review
REVIEW TYPE: Product Analysis
GRADE: Product Analysis, No Rating

REVISION DATE: 20030228

...reviewed to make the system hacker-proof by eliminating weaknesses exploited by hackers. A biomorphic **sequence** generator outputs random **numbers** for session IDS and security codes, and the system should be able to do so...
...Cisco Catalyst 6500 switches to allow customers to combine security with such IP services as **Voice** -over-IP (**VoIP**), wireless LAN integration, QoS (quality of service) features, and content switching.

16/3,K/4
DIALOG(R)File 256:SoftBase:Reviews,Companies&Prods.

00141129 DOCUMENT TYPE: Review

PRODUCT NAMES: OctiVox (130508)

TITLE: OctiVox
AUTHOR: Staff
SOURCE: Internet Telephony, v5 n8 p22(2) Aug 2002
ISSN: 1098-0008
HOMEPAGE: http://www.internettelephony.com

RECORD TYPE: Review
REVIEW TYPE: Review
GRADE: A

REVISION DATE: 20021230

Octiv's OctiVox improves **voice** -over-IP (**VoIP**) by reducing latency, eliminating echo, and improving **sound** quality. The developer's toolkit includes an API for performing **audio** processing that provides clarity and echo mitigation. The intelligibility enhancement function can improve conference room...
...distances. Octiv has also added DiamondWare's latency reduction algorithms into OctiVox for enhanced quality **VoIP** . OctiVox uses a buffer management system to reduce the roundtrip delay to as little as...

...gating to reduce apparent noise level and increase speaker clarity. In addition, OctiVox can perform **consonant** and vowel enhancement using an enhanced time/frequency model of the **speech** process.

DESCRIPTORS: Program Development; QoS (Quality of Service); **Sound** Processing; System Performance; **VoIP**

16/3,K/5
DIALOG(R)File 256:SoftBase:Reviews,Companies&Prods.

00131468 DOCUMENT TYPE: Review

PRODUCT NAMES: QoS (Quality of Service) (843954)

TITLE: 'Formal' QoS Not Yet A Priority: IT managers find simpler...
AUTHOR: Drucker, David
SOURCE: InternetWeek, v866 p22(1) Jun 18, 2001
ISSN: 0746-8121
HOMEPAGE: http://www.internetwk.com

RECORD TYPE: Review
REVIEW TYPE: Product Analysis
GRADE: Product Analysis, No Rating

REVISION DATE: 20020630

...QoS (quality of service), which will become more of a requirement when such technologies as **Voice** -over-IP (**VoIP**) and video are widely used. Although for many years QoS (Quality of Service) technologies for...

...network managers are currently turning to much more straightforward solutions. QoS assumes that because each **packet** of data is evaluated as it goes through a switch or a router, mechanisms can be created that identify priority **packets** and tell the network to deliver those specific **packets** before others. Creating enabling infrastructures has been difficult, and 'you really need to be a rocket scientist to understand how these **parameters** work and how they interact with each other across your network.' QoS can be practically...

...by using various queuing algorithms in routers and Layer 3 switches that when activated guide **packet** transmission priorities. Queuing methods are also called scheduling methods and use a standard called DiffServ...

16/3,K/6
DIALOG(R)File 256:SoftBase:Reviews,Companies&Prods.

00124756 DOCUMENT TYPE: Review

PRODUCT NAMES: Frame Relay (842729); Standards (830218)

TITLE: More users tuning in to voice **over frame relay**
AUTHOR: Rohde, David
SOURCE: Network World, v17 n16 p32(1) Apr 7, 2000
ISSN: 0887-7661
HOMEPAGE: http://www.nwfusion.com

RECORD TYPE: Review
REVIEW TYPE: Product Analysis
GRADE: Product Analysis, No Rating

REVISION DATE: 20011130

TITLE: More users tuning in to voice **over frame relay**

...fragmentation standard that is often used for any device dubbed a multiservice unit. Vendors of **voice** over frame relay use methods similar to those for **voice** over ATM, but less overhead is encountered. Frame relay is a spare, variable-length protocol...

...datagrams of 1,500 or more, as compared with ATM 53 in every cell. However, **voice** needs continuously flowing **packets** . All frame relay access devices assemble **voice** traffic in small, uniformly sized **packets** , even as data frames vary in length. With FRF.12, **voice** and data frames are broken into smaller fragments and a two-octet fragmentation header is added. The header includes a 112-binary-digit **sequence number** . Fragments have to arrive in order, or the transmission will drop **packets** . On a T1 or heftier frame relay link, fragmentation may not be an issue, since excess bandwidth may be available. However, **voice** over frame may, for example, require a branch office with a 56K or 128Kbit/sec frame relay connection to advance **voice** traffic across excess capacity. When that happens, users or a carrier have to package data and **voice** fragments in realistically sized **packets** , and must check the frame relay network **parameters** . Irrespective of the **voice** compression used, 120 percent of the compression rate will be required, but carriers generally provide separate PVCs for **voice** traffic. Among other topics covered are the FRF.8 interworking standard for data, FRF.11...

DESCRIPTORS: Communications Standards; Internetworking; Network Software; Standards; **VoIP**

16/3,K/7
DIALOG(R)File 256:SoftBase:Reviews,Companies&Prods.

00115592 DOCUMENT TYPE: Review

PRODUCT NAMES: Cognitel Windows 9x (742937)

TITLE: ID Callers with Versatile, Easy Voice **Mail Manager**

AUTHOR: Newman, Jeff
SOURCE: Windows Magazine, v10 n4 p46(1) Apr 1999
ISSN: 1060-1066
HOMEPAGE: http://www.winmag.com

RECORD TYPE: Review
REVIEW TYPE: Review
GRADE: A

REVISION DATE: 20030925

TITLE: ID Callers with Versatile, Easy Voice **Mail Manager**

Novcom's Cognitel, a robust desktop computer-telephony application, uses **voice** recognition to identify callers when Caller ID cannot. Cognitel also permits users to check **voice** mail messages from an e-mail program, as long as Microsoft Outlook or Exchange is used. **Voice** messages show in the inbox, with the caller's name listed as the subject of a message. Users can listen to **voice** messages, convert them to SAV files, and save for review later on. The user has...

...or personal greeting and records the message. Each time callers are identified, Cognitel adds new **voice** sample to the database. During testing, users imported an existing Outlook contact list and read off names to create **voice** samples. When the callers' voices were in the database, the recognition rate rose above 90 percent. The software uses real **voice** patterns instead of using **consonant** pronunciations for pattern recognition. Messages appeared in Outlook with **voice** message icons, and were easy to retrieve. A full-fledged TAPI-compliant **voice** modem is required, and although only a few are available, including the LT Win Modem ...

DESCRIPTORS: Computer Telephony; E-Mail; Exchange; IBM PC & Compatibles; Telecommunications; Telephone Messages; **Voice** Mail; Windows

16/3,K/8
DIALOG(R)File 256:SoftBase:Reviews,Companies&Prods.

00108415 DOCUMENT TYPE: Review

PRODUCT NAMES: Reality Windows 95 (709336)

TITLE: Sounds Sweet: Seer Systems' Reality
AUTHOR: Magel, Mark
SOURCE: AV Video & Multimedia Producer, v20 n4 p100(4) Apr 1998
ISSN: 1090-7459
HOMEPAGE: http://www.avvideo.com

RECORD TYPE: Review
REVIEW TYPE: Review
GRADE: A

REVISION DATE: 20040127

...already has an external synthesizer connected to the PC, it can play Reality as a **sound** module. There are three main screens in the interface: Bankset, Program, and Options. The Bankset window shows a program list of **sound** names, programs, **index numbers** , and other descriptors. Over a

dozen **sound** banks are included; each contains at least four dozen instruments that can be added to **sound** creations. The Options screen presents the general parameters of the Master Controls, and the Program...

DESCRIPTORS: IBM PC & Compatibles; MIDI; Multimedia; Music; **Sound** Processing; Windows

16/3,K/9
DIALOG(R)File 256:SoftBase:Reviews,Companies&Prods.

00104882 DOCUMENT TYPE: Review

PRODUCT NAMES: Emblaze Creator 2.01 Windows & Windows NT (674559)

TITLE: Emblaze Creator
AUTHOR: Sherman, Lee
SOURCE: NewMedia, v7 n11 p36(2) Sep 1, 1997
ISSN: 1060-7188
HOMEPAGE: http://www.newmedia.com

RECORD TYPE: Review
REVIEW TYPE: Review
GRADE: B

REVISION DATE: 20030527

...seem like a CD-ROM. The Emblaze file format permits real-time streaming animation, and **audio** and video at 12fps to 24fps over a standard 28.8Kbps connection. Emblaze's player...

...so that playback is greatly accelerated. Users need not concern themselves about downloading many different **audio** , video, and other multimedia plug-ins in order to view a site. However, Emblaze Creator...

...old-fashioned features, such as a basic drawing tool and a Timeline window that shows **sequence** frames only as **numbers** . The Data Monitor is a better tool because it tracks the data size and rate...
?

18/3,K/1
DIALOG(R)File 256:SoftBase:Reviews,Companies&Prods.

00122341 DOCUMENT TYPE: Review

PRODUCT NAMES: Digital Video (830268); Web Site Design (838543)

TITLE: Shooting Video for the Web
AUTHOR: Kelsey, Logan Feeley, Jim
SOURCE: Digital Video Magazine, p54(6) Feb 2000
ISSN: 1075-251X
HOMEPAGE: http://www.dv.com

RECORD TYPE: Review
REVIEW TYPE: Product Analysis
GRADE: Product Analysis, No Rating

REVISION DATE: 20010730

...easier to compress if the image within an image is kept simple and if each **frame** within a sequence closely resembles the **frame** before and after it. World Wide Web video has strict palette restrictions so strong colors...
...can solve one third of the problems associated with shooting Web video. High quality Web **audio** will make up for some of the deficiencies of Web video, and it is important to remember that the goal is to minimize extraneous noise to maximize **signal** -to- **noise** **ratio** .
?

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File   9:Business & Industry(R) Jul/1994-2004/Mar 15
         (c) 2004 Resp. DB Svcs.
File  15:ABI/Inform(R) 1971-2004/Mar 16
         (c) 2004 ProQuest Info&Learning
File  16:Gale Group PROMT(R) 1990-2004/Mar 16
         (c) 2004 The Gale Group
File  20:Dialog Global Reporter 1997-2004/Mar 16
         (c) 2004 The Dialog Corp.
File  47:Gale Group Magazine DB(TM) 1959-2004/Mar 16
         (c) 2004 The Gale group
File  75:TGG Management Contents(R) 86-2004/Mar W1
         (c) 2004 The Gale Group
File  80:TGG Aerospace/Def.Mkts(R) 1986-2004/Mar 16
         (c) 2004 The Gale Group
File  88:Gale Group Business A.R.T.S. 1976-2004/Mar 15
         (c) 2004 The Gale Group
File  98:General Sci Abs/Full-Text 1984-2004/Feb
         (c) 2004 The HW Wilson Co.
File 112:UBM Industry News 1998-2004/Jan 27
         (c) 2004 United Business Media
File 141:Readers Guide 1983-2004/Feb
         (c) 2004 The HW Wilson Co
File 148:Gale Group Trade & Industry DB 1976-2004/Mar 10
         (c)2004 The Gale Group
File 160:Gale Group PROMT(R) 1972-1989
         (c) 1999 The Gale Group
File 275:Gale Group Computer DB(TM) 1983-2004/Mar 16
         (c) 2004 The Gale Group
File 264:DIALOG Defense Newsletters 1989-2004/Mar 16
         (c) 2004 The Dialog Corp.
File 484:Periodical Abs Plustext 1986-2004/Mar W1
         (c) 2004 ProQuest
File 553:Wilson Bus. Abs. FullText 1982-2004/Feb
         (c) 2004 The HW Wilson Co
File 570:Gale Group MARS(R) 1984-2004/Mar 16
         (c) 2004 The Gale Group
File 608:KR/T Bus.News. 1992-2004/Mar 16
         (c)2004 Knight Ridder/Tribune Bus News
File 620:EIU:Viewswire 2004/Mar 12
         (c) 2004 Economist Intelligence Unit
File 613:PR Newswire 1999-2004/Mar 16
         (c) 2004 PR Newswire Association Inc
File 621:Gale Group New Prod.Annou.(R) 1985-2004/Mar 16
         (c) 2004 The Gale Group
File 623:Business Week 1985-2004/Mar 15
         (c) 2004 The McGraw-Hill Companies Inc
File 624:McGraw-Hill Publications 1985-2004/Mar 15
         (c) 2004 McGraw-Hill Co. Inc
File 634:San Jose Mercury  Jun 1985-2004/Mar 15
         (c) 2004 San Jose Mercury News
File 635:Business Dateline(R) 1985-2004/Mar 16
         (c) 2004 ProQuest Info&Learning
File 636:Gale Group Newsletter DB(TM) 1987-2004/Mar 16
         (c) 2004 The Gale Group
File 647:CMP  Computer Fulltext 1988-2004/Mar W1
         (c) 2004 CMP Media, LLC
File 696:DIALOG Telecom. Newsletters 1995-2004/Mar 15
         (c) 2004 The Dialog Corp.
File 674:Computer News Fulltext 1989-2004/Mar W1
         (c) 2004 IDG Communications
File 810:Business Wire 1986-1999/Feb 28
```

(c) 1999 Business Wire
File 813:PR Newswire 1987-1999/Apr 30
(c) 1999 PR Newswire Association Inc

Set	Items	Description
S1	6498978	VOICE OR AUDIO OR SOUND OR SPEECH
S2	2235861	FRAME?
S3	1517	(INTERPOLAT? OR ENCOD?)(5N)(REPEAT? OR ITERATIV? OR REDUNDANT? OR REITERA?)
S4	45	S3(10N)(RECOVER? OR ERROR?)
S5	17716	SIGNAL(3N)NOISE()RATIO
S6	222207	VOIP OR VOICE(3N)INTERNET
S7	84	S2(5N)(FIRST OR INITIAL?)(7N)(SECOND OR SUBSEQUENT?)(5N)ENCOD?
S8	877	PARAMETER?(5N)PACKET?
S9	17392	CONSONANT??
S10	32974	(INDEX OR SEQUENCE)(3N)NUMBER??
S11	0	S10(7N)(MULTIPLEX? OR MULTI()PLEX?)(5N)(TRANSMIT? OR TRANSMIS? OR SEND OR SENDING OR SENDS)
S12	40609	(SAME OR CLOSE OR EQUAL OR EQUIVALENT OR APPROXIMAT? OR MAXIMUM OR HIGHEST)(3N)MATCH?
S13	1	AU=(AMANO, F? OR AMANO F?)
S14	116949	(S1 OR S6)(S)S2
S15	3	S14(S)(S3 OR S4)
S16	2	RD S15 (unique items)
S17	40	S14(S)S5
S18	0	S17(S)S8
S19	0	S17(S)S9
S20	0	S17(S)S10
S21	0	S17(S)S12
S22	40	S17 NOT (S13 OR S15)
S23	8	S22 AND PY=2001:2004
S24	32	S22 NOT S23
S25	25	RD S24 (unique items)

13/3,K/1 (Item 1 from file: 88)
DIALOG(R)File 88:Gale Group Business A.R.T.S.

03656979 SUPPLIER NUMBER: 17221049
A multirate acoustic echo canceler structure.
Amano, Fumio ; Meana, Hector Perez; Luca, Adriano de; Duchen, Gonzalo
IEEE Transactions on Communications, v43, n7, p2172(5)
July, 1995
ISSN: 0090-6778 LANGUAGE: English RECORD TYPE: Citation

Amano, Fumio ...
?

16/3,K/1 (Item 1 from file: 20)
DIALOG(R)File 20:Dialog Global Reporter

29947201 (USE FORMAT 7 OR 9 FOR FULLTEXT)
Feature - Budget video editing - Movie making.
Tim Nott.
PC WORLD, p93
June 01, 2003
JOURNAL CODE: WPCW LANGUAGE: English RECORD TYPE: FULLTEXT
WORD COUNT: 5111

(USE FORMAT 7 OR 9 FOR FULLTEXT)

... decode and recode all the frames as editing is done. The lossiness of the re- **encoding** is progressive, so **repeated** editing in this format degrades the quality of the final product.
The third file format...

16/3,K/2 (Item 1 from file: 148)
DIALOG(R)File 148:Gale Group Trade & Industry DB

13298008 SUPPLIER NUMBER: 73023226 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Video improvements obviate big bit streams. (Technology Information)
Dipert, Brian
EDN, 46, 6, 83
March 15, 2001
ISSN: 0012-7515 LANGUAGE: English RECORD TYPE: Fulltext
WORD COUNT: 6158 LINE COUNT: 00556

... reduction tends to eliminate fine image detail. Any good lossy video encoder automatically discards redundant **frame** -to- **frame** information, so an inverse- telecine filter may not dramatically reduce the compressed bit rate, but...

...characteristics of the source material. VBR (variable-bit-rate) video encoding, as with VBR lossy **audio** , enables encoders to intelligently allocate bits across **frames** as necessary (Reference A).
RealVideo 8 introduced a new video codec that RealNetworks based on
...
?

25/3,K/1 (Item 1 from file: 9)
DIALOG(R)File 9:Business & Industry(R)

2738881 Supplier Number: 02738881 (USE FORMAT 7 OR 9 FOR FULLTEXT)
JVC Gives Details On DVD A/V Deck
(JVC is introducing its new DVD Video/Audio deck, Super VHS, and analog TV, but it is not saying when it will enter such categories as recordable DVD Video, D-VHS, and Internet audio)
TWICE, v 15, n 5, p 8
February 21, 2000
DOCUMENT TYPE: Journal ISSN: 0892-7278 (United States)
LANGUAGE: English RECORD TYPE: Fulltext
WORD COUNT: 429

(USE FORMAT 7 OR 9 FOR FULLTEXT)

TEXT:
...As for the categories it did talk about, the XV-D723GD is a DVD Video/ **Audio** deck, which will be available in June at an $899.95, suggested retail. On the...

...high-bit/high-sampling video D/A converter, 500 lines of horizontal resolution, and a **signal** -to- **noise** **ratio** of 65dB. JVC's direct progressive scan turns the 24- **frames** per second format (fps) of film into 60 fps for TV display without any intervening...

25/3,K/2 (Item 1 from file: 15)
DIALOG(R)File 15:ABI/Inform(R)

00917320 95-66712
Test approaches to GSM
Rosar, Werner
Telecommunications (International Edition) v28n8 PP: 53-56 Aug 1994
JRNL CODE: TIE
WORD COUNT: 2814

...TEXT: measured values behave in relation to time. Unlike classic radio testing in which a certain **signal** -to- **noise** **ratio** was measured at the **audio** output of the receiver, the GSM receiver test is concerned with whether the received bit signal is free of errors. The significant parameters are the bit error rate (BER) or **frame** erasure ratio (FER) which are determined by transmitting a known, long pseudo-random bit stream ...

25/3,K/3 (Item 1 from file: 16)
DIALOG(R)File 16:Gale Group PROMT(R)

07339979 Supplier Number: 61792903 (USE FORMAT 7 FOR FULLTEXT)
new products.
Lasers & Optronics, v19, n3, p10
March, 2000
Language: English Record Type: Fulltext
Document Type: Tabloid; Academic Trade
Word Count: 7721

... at various speeds without smearing. Its analog signal has been carefully processed to achieve optimum **signal** -to- **noise** **ratio** by using Correlated Double Sampling. The camera's advanced asynchronous trigger function can allow input of a random external trigger pulse for instant video output as well. **Audio** Visual Supply, 4575 Ruffner Street, San Diego, CA 92111.
* Write in 228 or Reply Online...

25/3,K/4 (Item 2 from file: 16)
DIALOG(R)File 16:Gale Group PROMT(R)

05283188 Supplier Number: 48046898 (USE FORMAT 7 FOR FULLTEXT)
Filter implements CDMA, AMPS specs
Burt, Andrew
Electronic Engineering Times, p90
Oct 13, 1997
Language: English Record Type: Fulltext
Document Type: Magazine/Journal; Trade
Word Count: 1086

... characteristics of the CDMA path I/Q filters have a major impact on the input **signal** -to- **noise** **ratio** . The **Frame** Error Rate (FER) from the demodulation algorithm must be low enough to avoid **speech** breakup when the digital signal is reconstructed. The receive path's in-band amplitude and...

25/3,K/5 (Item 3 from file: 16)
DIALOG(R)File 16:Gale Group PROMT(R)

04177990 Supplier Number: 46103890 (USE FORMAT 7 FOR FULLTEXT)
SONY'S NEW HIGH RESOLUTION CAMERA MODULE INCORPORATES ADVANCED HIGH-SPEED PARTIAL SCANNING
News Release, pN/A
Jan 30, 1996
Language: English Record Type: Fulltext
Document Type: Magazine/Journal; Trade
Word Count: 760

(USE FORMAT 7 FOR FULLTEXT)
TEXT:
...offers some distinct advantages over conventional interlaced CCD cameras, such as electronic shuttering a full **frame** of video (both odd and even fields) simultaneously and outputting the video lines sequentially. This...

...Because only a limited number of lines are used, the camera can output a partial **frame** of video up to four times faster than using the total vertical resolution. The result...

...and efficiency are crucial. With its dual video outputs, the XC-8500CE provides a full **frame** of image capture at 50 Hz **frame** rate, twice the speed of interlaced cameras. Three different output modes are available: non-interlaced video at 25Hz and 50Hz **frame** rates, and interlaced video at 50 Hz. In addition, the camera incorporates Sony's proprietary...

...camera's high-speed partial scan or E-Donpisha modes. This optional

adapter provides two **frames** of memory to display the camera's progressive scan images on CCIR, EIA or multi...

...in superb light sensitivity of 3 Lux at FI.4. Incorporating square pixels and enhanced **signal** -to- **noise** **ratio** of 58 dB, the camera is ideally suited for measurement applications. It offers a number of other features including: full- **frame** electronic shutter speeds from 1/125 to 1/100,000 per second, long exposure mode...

...Professional Products Group of Sony Electronics is a leading U.S. supplier of video and **audio** equipment for the broadcast, production, business, industry, government, medical and education markets. Sony offers a...

25/3,K/6 (Item 4 from file: 16)
DIALOG(R)File 16:Gale Group PROMT(R)

03948192 Supplier Number: 45715948 (USE FORMAT 7 FOR FULLTEXT)
PANASONIC DVC CAMCORDER DUE HERE IN OCT.
Consumer Electronics, v35, n32, pN/A
August 7, 1995
Language: English Record Type: Fulltext
Document Type: Newsletter; Trade
Word Count: 524

... line picture, "nearly 20% more than laserdisc and 50% more than a live TV broadcast." **Signal** -to- **noise** **ratio** is 54 dB, said to be 2-3 times better than existing consumer video equipment. Panasonic model can record **audio** in 2-channel 16-bit version or 2 sets of 12-bit (Sony's uses ...

...optical and 20:1 digital zoom, digital image stabilization. Camcorder can be used for still- **frame** recording, providing 580 6-sec. shots on 60-min. cassette while normal **audio** continues.
Dimensions and weight of Panasonic camcorder weren't given, but Matsushita says its model...

25/3,K/7 (Item 5 from file: 16)
DIALOG(R)File 16:Gale Group PROMT(R)

03056614 Supplier Number: 44159652
A new look
Network World, p41
Oct 11, 1993
Language: English Record Type: Abstract
Document Type: Magazine/Journal; Trade

ABSTRACT:
...user's needs. A table lists information such as system and bus type, standards, compression, **frame** rate, **signal** -to- **noise** **ratio** and price for 20 products from 9 companies. Short list with brief summaries of 4...

...interoperability standards are discussed. Another table displays information on video-to-desktop options. Differences in **audio** and video quality are outlined, and system upgrading considerations are also discussed. ...

25/3,K/8 (Item 1 from file: 20)
DIALOG(R)File 20:Dialog Global Reporter

02959458
Hybrid Networks to Support SpeedChoice Launch of High-Speed Wireless Data Services
PR NEWSWIRE
September 29, 1998
JOURNAL CODE: WPRW LANGUAGE: English RECORD TYPE: FULLTEXT
WORD COUNT: 639

... from 256Kb to 5Mbps, telephone return up to 33.6Kbps, or router return via ISDN, **Frame** Relay or T1. Hybrid Networks splits the 6Mhz downstream channel into three 2MHz subchannels each...

... uncertainties. Hybrid Networks, Inc. 6409 Guadalupe Mines Rd. San Jose, CA 95120 408-323-6500 **voice** 408-323-6471 fax info@hybrid.com http://www.hybrid.com /CONTACT: Sandra DeRodeff of...

25/3,K/9 (Item 2 from file: 20)
DIALOG(R)File 20:Dialog Global Reporter

01280711
NCTI Completes Licensing Agreements for Art Gekko Line of Printed Speaker Grilles
BUSINESS WIRE
March 30, 1998 14:2
JOURNAL CODE: WBWE LANGUAGE: English RECORD TYPE: FULLTEXT
WORD COUNT: 312

... The Company specializes in the utilization of sound and signal waves to reduce noise, improve **signal** -to- **noise** **ratio** and enhance **sound** quality. For more information, refer to the Company's World Wide Web site at http:

25/3,K/10 (Item 1 from file: 47)
DIALOG(R)File 47:Gale Group Magazine DB(TM)

04796912 SUPPLIER NUMBER: 17438392 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Digital camcorders arrive. (digital video cassette camcorders)
Electronics Now, v66, n11, p6(1)
Nov, 1995
ISSN: 1067-9294 LANGUAGE: English RECORD TYPE: Fulltext; Abstract
WORD COUNT: 562 LINE COUNT: 00047

... as "nearly 20% more than laserdisc and 50% more than a live broadcast." Panasonic's **signal** -to- **noise** **ratio** is given as 54 dB, two to three times better than existing consumer video equipment. All versions contain a "snapshot" mode, permitting them to be used as high-resolution, full- **frame** still cameras while normal **audio** is continued. Sony's models have two 12-bit stereo **audio** pairs, while the Panasonic model offers two **audio** modes--12-bit for two soundtracks and 16-bit for a single stereo pair.

Obviously...

25/3,K/11 (Item 2 from file: 47)
DIALOG(R)File 47:Gale Group Magazine DB(TM)

02440572 SUPPLIER NUMBER: 02865515 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Compact disc digital audio systems; how the new digital audio 4.7-inch playback record system works.
Ranada, David
Computers & Electronics, v21, p41(8)
Aug, 1983
ISSN: 0745-1458 LANGUAGE: ENGLISH RECORD TYPE: FULLTEXT
WORD COUNT: 6039 LINE COUNT: 00455

... and occur 7,350 times per second-once per frame--a voice signal with a **signal** -to- **noise** **ratio** of about 45 dB and a frequency range up to about 3,500 Hz can...

25/3,K/12 (Item 1 from file: 88)
DIALOG(R)File 88:Gale Group Business A.R.T.S.

05391703 SUPPLIER NUMBER: 61487481
Separation of Speech from Interfering Sounds Based on Oscillatory Correlation.
Wang, DeLiang L.; Brown, Guy J.
IEEE Transactions on Neural Networks, 10, 3, 684
May, 1999
ISSN: 1045-9227 LANGUAGE: English RECORD TYPE: Abstract

AUTHOR ABSTRACT: A multistage neural model is proposed for an auditory scene analysis task--segregating **speech** from interfering **sound** sources. The core of the model is a two-layer oscillator network that performs stream segregation on the basis of oscillatory correlation. In the oscillatory correlation **framework** , a stream is represented by a population of synchronized relaxation oscillators, each of which corresponds...

...auditory representations are formed. The model has been systematically evaluated using a corpus of voiced **speech** mixed with interfering sounds, and produces improvements in terms of **signal** -to- **noise** **ratio** for every mixture. The performance of our model is compared with other studies on computational...

25/3,K/13 (Item 1 from file: 141)
DIALOG(R)File 141:Readers Guide

03083501 H.W. WILSON RECORD NUMBER: BRGA95083501 (USE FORMAT 7 FOR FULLTEXT)
Digital camcorders arrive.
Lachenbruch, David.
Electronics Now (Electron Now) v. 66 (Nov. '95) p. 6
WORD COUNT: 535

(USE FORMAT 7 FOR FULLTEXT)

TEXT:
... 20[percent] more than laserdisc and 50[percent] more than a live broadcast." Panasonic's **signal** -to- **noise** **ratio** is given as 54 dB, two to three times better than existing consumer video equipment. All versions contain a "snapshot" mode, permitting them to be used as high-resolution, full- **frame** still cameras while normal **audio** is continued. Sony's models have two 12-bit stereo **audio** pairs, while the Panasonic model offers two **audio** modes--12-bit for two soundtracks and 16-bit for a single stereo pair.
Obviously...

25/3,K/14 (Item 2 from file: 141)
DIALOG(R)File 141:Readers Guide

02273630 H.W. WILSON RECORD NUMBER: BRGA92023630
Proscan combi player.
AUGMENTED TITLE: PSLD41
Video v. 16 (Apr. 1992) p. 13+

ABSTRACT: Ease of use and good video and **audio** performance are hallmarks of the ProScan PSLD41 combination laser disc and CD player ($899). The PSLD41 offers video noise reduction, a peak **audio** level search button, and a special CD mode that produces a purer **sound** from CDs. The machine's only disadvantages are its lack of digital effects for producing still **frames** , slow motion, and speed play from most laser discs and its weakness in the chroma PM **signal** -to- **noise** **ratio** , which is a common fault among combination players.

25/3,K/15 (Item 1 from file: 148)
DIALOG(R)File 148:Gale Group Trade & Industry DB

09332235 SUPPLIER NUMBER: 19160440 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Image recording: video quality enough for sciences and arts. (QuVIS' QuBit family of image recorders)
Goertzen, Kenbe
Advanced Imaging, v12, n1, p34(3)
Jan, 1997
ISSN: 1042-0711 LANGUAGE: English RECORD TYPE: Fulltext
WORD COUNT: 2319 LINE COUNT: 00187

... range of throughput required. Each allows control of frame rate, pixel rate, frame size, bandwidth, **signal** -to- **noise** **ratio** and interlaced or non-interlaced operation. All configurations support four-channel 24-bit digital or analog **audio** IO. The base QuBit is intended to address

25/3,K/16 (Item 1 from file: 160)
DIALOG(R)File 160:Gale Group PROMT(R)

01473289
Marantz Japan.
DEMPA DIGEST October 6, 1986 p. 9

Marantz has introduced a lower priced laser optical video disc player. The LV101 features digital **audio** circuits and has 400 horizontal line resolution and 45 decibel video **signal** -to- **noise** **ratio** . It is capable of still and **frame** forward reproduction with a companion remote control. ...

25/3,K/17 (Item 2 from file: 160)
DIALOG(R)File 160:Gale Group PROMT(R)

00477462
Panasonic (Secaucus, NJ) has introduced a solenoid-operated video cassette recorder/player and a solenoid operated-video cassette player.
News Release (for further information apply to company indexed) April, 1979 p. 1,2

Both the Omnivision II NV-8200 recorder/player and the NV-8170 feature 2 separate **audio** channels, unattended auto-repeat, playback speeds of 20-150% and a 2x speed with **audio** and still **frame** and single advance. The NV-8200 also offers color or black-and-white recording and...

... with Panasonic NV-T120 cassettes (with 30 and 60 min cassettes available), a 45 dB **signal** / **noise** **ratio** , 300 lines monochrome and 240 lines color resolution, direct drive video head cylinder motor, and...

25/3,K/18 (Item 1 from file: 275)
DIALOG(R)File 275:Gale Group Computer DB(TM)

01889152 SUPPLIER NUMBER: 17813445
Digital video's new age. (the Digital Video Cassette standard and the Sony DCR-VX1000 Digital Handycam)(First Look)(Product Announcement)
Waring, Becky
Newmedia, v6, n1, p17(1)
Jan 2, 1996
DOCUMENT TYPE: Product Announcement ISSN: 1060-7188 LANGUAGE: English RECORD TYPE: Abstract

...ABSTRACT: 25,000 system could in 1995. Features of the standard include 500-line resolution, 54dB **signal** -to- **noise** **ratio** , separate video and PCM stereo or quad **audio** tracks, 5-to-1 compression, a still- **frame** mode, and support for forthcoming HDTV and wide-screen modes. DVC applications will be easy...

25/3,K/19 (Item 1 from file: 484)
DIALOG(R)File 484:Periodical Abs Plustext

02603527 (USE FORMAT 7 OR 9 FOR FULLTEXT)
Digital camcorders arrive
Lachenbruch, David
Electronics Now (GRAD), v66 n11, p6
Nov 1995
ISSN: 1067-9294 JOURNAL CODE: GRAD
DOCUMENT TYPE: News
LANGUAGE: English RECORD TYPE: Fulltext; Abstract
WORD COUNT: 493 LENGTH: Medium (10-30 col inches)

TEXT:
... as "nearly 20% more than laserdisc and 50% more than a live broadcast." Panasonic's **signal** -to- **noise** **ratio** is given as 54 dB, two to three times better than existing consumer video equipment. All versions contain a "snapshot" mode, permitting them to be used as high-resolution, full- **frame** still cameras while normal **audio** is continued. Sony's models have two 12-bit stereo **audio** pairs, while the Panasonic model offers two **audio** modes--12-bit for two soundtracks and 16-bit for a single stereo pair.

Obviously...

25/3,K/20 (Item 1 from file: 624)
DIALOG(R)File 624:McGraw-Hill Publications

0106789
Board Supports Video/Data/Voice
; Pg 72; Vol. 14, No. 1
Section Heading: What's News: Graphics
Word Count: 239 *Full text available in Formats 5, 7 and 9*

TEXT:
... The dynamic range is 78 decibels, and the bandwidth is either 3.4 kHz for **voice** or 11 kHz for higher fidelity. **Signal** -to- **noise** **ratio** is 40 decibels, and the digital data is 6032 bytes per **frame** or 325 megabytes per disk. Data transfer is rated at 1.45 megabits per second...

25/3,K/21 (Item 1 from file: 636)
DIALOG(R)File 636:Gale Group Newsletter DB(TM)

01015984 Supplier Number: 40359508 (USE FORMAT 7 FOR FULLTEXT)
NEW TV/VCR LINES
Consumer Electronics, pN/A
April 18, 1988
Language: English Record Type: Fulltext
Document Type: Newsletter; Trade
Word Count: 930

... sequence, has 8-bit digital field memory, making possible such special effects as still with **sound** , strobe motion with **sound** and freeze **frame** for both CLV and CAV discs. It has horizontal resolution of 425 lines, 46dB **signal** -to- **noise** **ratio** . To be available in June, it will be $1,700.

Proton debuts 20" flat square...

25/3,K/22 (Item 1 from file: 647)
DIALOG(R)File 647:CMP Computer Fulltext

01141360 CMP ACCESSION NUMBER: EET19971013S0074
Filter implements CDMA, AMPS specs
Andrew Burt, Product Manager, Wireless Systems Group, GEC Plessey Semiconductors Inc., Scotts Valley, Calif.
ELECTRONIC ENGINEERING TIMES, 1997, n 975, PG90
PUBLICATION DATE: 971013

JOURNAL CODE: EET   LANGUAGE: English
RECORD TYPE: Fulltext
SECTION HEADING: Analog/Mixed-Signal Design
WORD COUNT: 1080

...   characteristics of the CDMA path I/Q filters have a major impact on the input **signal** -to- **noise** **ratio** . The **Frame** Error Rate (FER) from the demodulation algorithm must be low enough to avoid **speech** breakup when the digital signal is reconstructed. The receive path's in-band amplitude and...

25/3,K/23   (Item 1 from file: 696)
DIALOG(R)File 696:DIALOG Telecom. Newsletters

00730114
What's Making Waves In The Market
Electronic Commerce News
June 12, 2000 VOL: 5   ISSUE: 24   DOCUMENT TYPE: NEWSLETTER
PUBLISHER: PHILLIPS BUSINESS INFORMATION
LANGUAGE: ENGLISH   WORD COUNT: 770   RECORD TYPE: FULLTEXT

TEXT:
...s protocol and bandwidth flexibility
-- supporting both inverse multiplexing over ATM (IMA) and multi-link
frame
relay (MFR) capabilities on a port-by-port basis--with a high-density
interface
for...

...in the central office
or POP, the new A-3010 combines the functionality of a **frame** relay
switch, an
ATM switch, M1/3 mux and has 1/0 DACS capabilities. While...
...DS-3
including: backhauling connections to the service-appropriate edge switch,
maintaining and provisioning legacy **voice** -oriented central office
equipment and
managing space constraints in the central office. The A-3010...

...application that provides the quickest method to measure and record
channel
power, wavelength and optical **signal** -to- **noise** **ratio** (OSNR). Agilent's
WDM
application uses a dual sweep technique in which one sweep uses...

25/3,K/24   (Item 2 from file: 696)
DIALOG(R)File 696:DIALOG Telecom. Newsletters

00715522
APEX DVD DECK DEFEATS ANTICOPY CODING, OUR LAB TEST SHOWS
AUDIO WEEK
March 6, 2000 DOCUMENT TYPE: NEWSLETTER
PUBLISHER: WARREN PUBLISHING INC.
LANGUAGE: ENGLISH   WORD COUNT: 1948   RECORD TYPE: FULLTEXT

TEXT:
...for $149-$199, depending on CC store. Other features include S-Video output; digital coaxial **audio** output for Dolby Digital, DTS or MPEG-2 **audio** ; Video CD (VCD) and Super VCD playback including karaoke discs, for which deck has 2...

...would have made Apex player popular, although our lab measurements reveal just average video and **audio** performance. But interest in deck skyrocketed when reports appeared on Internet touting "secret menu" that...
...we tried failed to play correctly without CSS decryption. Typical symptoms of scrambling included stuttered **audio** accompanied either by no picture or by fragmented and pixilated images. These, like occasional clear **frame** , ultimately froze onscreen --and image wouldn't change when movies were advanced to subsequent chapters of VCDs or **audio** CDs of any type, including MP3-compressed discs.

Rationale for turning off CSS encryption is...Barr, APEL pres. and veteran CE engineer involved with industry-standard measurements for video and **audio** performance.

Apex deck was middling in video frequency response, which measures resolution or sharpness of...

...red -- not veering toward magenta or yellow -- but not dead-on bull's-eye.

Video **signal** -to- **noise** **ratio** was just fair compared with other decks. In test for luminance (B&W) noise, Apex...
...Essentials and Sony HLX-4001 for video, CBS CD-1 and Pierre Verany discs for **audio** . Display monitor used for visual evaluation was 27" Toshiba CN27H95 with component video inputs.

Dicier...nearly identical clone of those for earlier Toshiba SD-2108 and SD-3109 DVD players.

Audio numbers for Apex were on par with other Chinese decks, though not as good as...

...of other Chinese decks. Rolloff had been fraction of dB in decks previously measured. In **audio** **signal** -to- **no**ise ratio , Apex weighed in at 89.3 dB, and dynamic range measured was 91.1 dB...

...channels of stereo signal. Good measurement also ensures accurate steering among channels in Dolby surround **sound** modes. Previously, best separation measured was 91 dB.

In hands-on evaluation, Apex compared favorably...
...Drive from ESS Technologies, Fremont, Cal. Latter said that besides decoding MPEG-2 video and **audio** , Dolby Digital

and MPEG-1 VCDs, its Swan-DVD Solution chip controls DVD navigation and...for $149-$199, depending on CC store. Other features include S-Video output; digital coaxial **audio** output for Dolby Digital, DTS or MPEG-2 **audio** ; Video CD (VCD) and Super VCD playback including karaoke discs, for which deck has 2...

...would have made
Apex player popular, although our lab measurements reveal just average video and **audio** performance. But interest in deck skyrocketed when reports appeared on Internet touting "secret menu" that...

...we tried failed to play correctly without CSS decryption. Typical symptoms of scrambling included stuttered **audio** accompanied either by no picture or by fragmented and pixilated images. These, like occasional clear **frame** , ultimately froze onscreen --and image wouldn't change when movies were advanced to subsequent chapters...
...correctly. As expected, setting of CSS function had no effect on performance of VCDs or **audio** CDs of any type, including MP3-compressed discs.

Rationale for turning off CSS encryption is...Barr, APEL pres. and veteran CE engineer involved with industry-standard measurements for video and **audio** performance.

Apex deck was middling in video frequency response, which measures resolution or sharpness of...

...red --
not veering toward magenta or yellow -- but not dead-on bull's-eye.

Video **signal** -to- **noise** **ratio** was just fair compared with other decks. In test for luminance (B&W) noise, Apex...

...Essentials and Sony
HLX-4001 for video, CBS CD-1 and Pierre Verany discs for **audio** . Display monitor used for visual evaluation was 27" Toshiba CN27H95 with component video inputs.

Dicier...nearly identical clone of those for earlier Toshiba SD-2108 and SD-3109 DVD players.

Audio numbers for Apex were on par with other Chinese decks, though not as good as...

...of other Chinese
decks. Rolloff had been fraction of dB in decks previously measured. In **audio** **signal** -to- **noise** **ratio** , Apex weighed in at 89.3 dB, and dynamic range measured was 91.1 dB...

...channels of stereo signal. Good
measurement also ensures accurate steering among channels in Dolby surround **sound** modes. Previously, best separation measured was 91 dB.

In hands-on evaluation, Apex compared favorably...

...Drive from ESS Technologies, Fremont, Cal. Latter
said that besides decoding MPEG-2 video and **audio** , Dolby Digital
and MPEG-1 VCDs, its Swan-DVD Solution chip controls DVD
navigation and...

25/3,K/25 (Item 3 from file: 696)
DIALOG(R)File 696:DIALOG Telecom. Newsletters

00599092
THIS WEEK IN MULTIMEDIA HARDWARE
MULTIMEDIA WEEK
April 6, 1998 VOL: 7 ISSUE: 14 DOCUMENT TYPE: NEWSLETTER
PUBLISHER: PHILLIPS BUSINESS INFORMATION
LANGUAGE: ENGLISH WORD COUNT: 317 RECORD TYPE: FULLTEXT

TEXT:
...Based on 3Dfx Voodoo Graphics technology, board features 6 MB of
memory, a 2 MB **frame** buffer for shapes and scenes and 4 MB for texture
processing. This less expensive version...

...a TV-out capability. Available Now

Turtle Beach Systems (http://www.tbeach.com)
Montego A3Dxstream **audio** card
Platform: PC Cost: $129
Contact: Mike McDougall Phone: 716/288-6900
Features 64- **voice** sounds that are delivered through 18-bit D/A
converters; includes a high-speed, bus-mastering PCI **audio** interface
and a 92 dB **signal** -to- **noise** **ratio** ; compatible with AC'97, PC'97 and
PC'98 specs; supports legacy DOS-based games...

...JPEG lossless compression; 130
real-time transitions, 3-track compositing and titling; supports real-
time **audio** playbacks of six or more stereo channels. Available now.

NEC Technologies Inc. (http://www.nec...
?

File 348:EUROPEAN PATENTS 1978-2004/Mar W01
(c) 2004 European Patent Office
File 349:PCT FULLTEXT 1979-2002/UB=20040311,UT=20040304
(c) 2004 WIPO/Univentio

Set	Items	Description
S1	155414	VOICE OR AUDIO OR SOUND OR SPEECH
S2	287032	FRAME?
S3	4600	(INTERPOLAT? OR ENCOD?)(5N)(REPEAT? OR ITERATIV? OR REDUND-ANT? OR REITERA?)
S4	276	S3(10N)(RECOVER? OR ERROR?)
S5	23043	SIGNAL(3N)NOISE()RATIO
S6	3322	VOIP OR VOICE(3N)INTERNET
S7	1598	S2(5N)(FIRST OR INITIAL?)(7N)(SECOND OR SUBSEQUENT?)(5N)EN-COD?
S8	2074	PARAMETER?(5N)PACKET?
S9	1345	CONSONANT??
S10	31479	(INDEX OR SEQUENCE)(3N)NUMBER??
S11	24	S10(7N)(MULTIPLEX? OR MULTI()PLEX?)(5N)(TRANSMIT? OR TRANS-MIS? OR SEND OR SENDING OR SENDS)
S12	10770	(SAME OR CLOSE OR EQUAL OR EQUIVALENT OR APPROXIMAT? OR MA-XIMUM OR HIGHEST)(3N)MATCH?
S13	6697	IC=G10L?
S14	8	(S1 OR S6)(7N)S2(S)S4
S15	0	S14(S)S8
S16	0	S14(S)S9
S17	3	S14(S)S10
S18	0	S14(S)S11
S19	0	S14(S)S12
S20	3	S14(S)S5
S21	0	S20 NOT S17
S22	5	S14 NOT S20
S23	0	S1(5N)S9(10N)S12
S24	598	S1(S)S9
S25	0	S24(10N)S8
S26	1	S24(S)S3
S27	1	S26 NOT (S14 OR S20)
S28	3	S24(S)S10
S29	3	S28 NOT (S26 OR S14 OR S20)
S30	0	S24(10N)S7
S31	2	S24(S)S12
S32	2	S31 NOT (S28 OR S26 OR S14 OR S20)
S33	124	(S1 OR S6)(5N)S7
S34	2	S33(10N)(S3 OR S4)
S35	2	S34 NOT (S31 OR S28 OR S26 OR S14 OR S20)
S36	2	S33(10N)S5
S37	2	S36 NOT (S34 OR S31 OR S28 OR S26 OR S14 OR S20)
S38	0	S33(10N)S8
S39	0	S33(10N)S9
S40	0	S33(10N)S10

17/3,K/1 (Item 1 from file: 348)
DIALOG(R)File 348:EUROPEAN PATENTS

01324138
Apparatus and method for digital data transmission
Vorrichtung und Verfahren zur digitalen Datenubertragung
Dispositif et procede de transmission de donnees numeriques
PATENT ASSIGNEE:
  Terayon Communication Systems, Inc., (2769080), 2952 Bunker Hill Lane,
    Santa Clara, CA 95054, (US), (Applicant designated States: all)
INVENTOR:
  Rakib, Selim Shlomo, Dr., 10271 West Acres, Cupertino, California 95014,
    (US)
  Azenkot, Yehuda, 1128 Littleoak Circle, San Jose, California 95129, (US)
LEGAL REPRESENTATIVE:
  Brax, Matti Juhani (85201), Berggren Oy Ab, P.O. Box 16, 00101 Helsinki,
    (FI)
PATENT (CC, No, Kind, Date): EP 1130919 A2 010905 (Basic)
                             EP 1130919 A3 020410
APPLICATION (CC, No, Date): EP 2001104541 960725;
PRIORITY (CC, No, Date): US 519630 950825; US 588650 960119; US 684243
    960719
DESIGNATED STATES: BE; DE; FR; GB; IE; NL
RELATED PARENT NUMBER(S) - PN (AN):
  EP 858695 (EP 96927270)
INTERNATIONAL PATENT CLASS: H04N-007/173; H04L-012/28; H04J-011/00;
  H04J-013/02; H04J-003/06; H04B-001/707; H04L-005/02; H04L-027/38
ABSTRACT WORD COUNT: 143
NOTE:
  Figure number on first page: 49

LANGUAGE (Publication,Procedural,Application): English; English; English
FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS A	(English)	200136	5384
SPEC A	(English)	200136	67833
Total word count - document A			73217
Total word count - document B			0
Total word count - documents A + B			73217

...SPECIFICATION is used to distinguish between signals from different cells. Variable rate is used on the **voice** channel to prevent transmissions when there is no meaningful data to send. All cells in... deshuffler circuit. All shuffler and deshuffler circuits receive the same seed and generate the same **sequence** of pseudorandom **numbers** therefrom. These pseudorandom numbers are used to generate read pointers to a framer memory and...

17/3,K/2 (Item 2 from file: 348)
DIALOG(R)File 348:EUROPEAN PATENTS

01291322
Apparatus and method for digital data transmission
Vorrichtung und Verfahren zur digitalen Datenubertragung
Procede et dispositif de transmission de donnees numeriques
PATENT ASSIGNEE:
  Terayon Communication Systems, Inc., (2769080), 2952 Bunker Hill Lane,

Santa Clara, CA 95054, (US), (Applicant designated States: all)
INVENTOR:
Rakib, Selim Shlomo, Dr., 10271 West Acres,, Cupertino, California 95014, (US)
Azenkot, Yehuda, 1128 LIttleoak Circle, San Jose, California 95129, (US)
LEGAL REPRESENTATIVE:
Brax, Matti Juhani (85201), Berggren Oy Ab, P.O. Box 16, 00101 Helsinki, (FI)
PATENT (CC, No, Kind, Date): EP 1107599 A2 010613 (Basic)
EP 1107599 A3 020508
APPLICATION (CC, No, Date): EP 2001104543 960725;
PRIORITY (CC, No, Date): US 519630 950825; US 588650 960119; US 684243 960719
DESIGNATED STATES: BE; DE; FR; GB; IE; NL
RELATED PARENT NUMBER(S) - PN (AN):
EP 858695 (EP 96927270)
INTERNATIONAL PATENT CLASS: H04N-007/173; H04L-012/28; H04J-011/00; H04J-013/02; H04J-003/06; H04B-001/707; H04L-005/02
ABSTRACT WORD COUNT: 143
NOTE:
Figure number on first page: 49

LANGUAGE (Publication,Procedural,Application): English; English; English
FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS A	(English)	200124	1478
SPEC A	(English)	200124	67821
Total word count - document A			69299
Total word count - document B			0
Total word count - documents A + B			69299

...SPECIFICATION is used to distinguish between signals from different cells. Variable rate is used on the **voice** channel to prevent transmissions when there is no meaningful data to send. All cells in... high susceptibility of QPSK modulation to narrowband interference. Narrowband interference results when a signal like **Voice** of America or a harmonic which has a bandwidth similar to the bandwidth of the...of the invention

In its first embodiment the invention provides for a Trellis encoder for **encoding** payload data bits with **redundant** bits and mapping the resulting bits into a constellation point. The Trellis encoder comprises the...

...recited in claim 1.

In its second embodiment the invention provides for an encoder for **encoding** payload data bits with **redundant** bits and mapping the resulting bits into a constellation point so as to achieve a...deshuffler circuit. All shuffler and deshuffler circuits receive the same seed and generate the same **sequence** of pseudorandom **numbers** therefrom. These pseudorandom numbers are used to generate read pointers to a framer memory and...

17/3,K/3 (Item 1 from file: 349)
DIALOG(R)File 349:PCT FULLTEXT

00368534 **Image available**
APPARATUS AND METHOD FOR DIGITAL DATA TRANSMISSION
DISPOSITIF ET PROCEDE DE TRANSMISSION DE DONNEES NUMERIQUES
Patent Applicant/Assignee:

TERAYON CORPORATION,
Patent and Priority Information (Country, Number, Date):
Patent: WO 9708861 A1 19970306
Application: WO 96US12391 19960725 (PCT/WO US9612391)
Priority Application: US 95519630 19950825; US 96588650 19960119
Designated States: AL AM AT AU AZ BB BG BR BY CA CH CN CZ DE DK EE ES FI GB GE HU IL IS JP KE KG KP KR KZ LK LR LS LT LU LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK TJ TR TT UA UG UZ VN KE LS MW SD SZ UG AT BE CH DE DK ES FI FR GB GR IE IT LU MC NL PT SE BF BJ CF CG CI CM GA GN ML MR NE SN TD TG
Publication Language: English
Fulltext Word Count: 91760

Fulltext Availability:
Detailed Description

Detailed Description
... networks such as ATM or ISDN that are designed for delivery of digitized video, digitized **audio** and digital data over point to point LAN connections.

Thus, the a major problem exists...can suddenly "pop up" when a subscriber turns on his or her TV or when **Voice** of America starts broadcasting. This sudden pop-up interference can jam a channel thereby causing...shuffler 3 0 and deshuff ler circuits receive the same seed and generate the same **sequence** of pseudorandom **numbers** therefrom. These pseudorandom numbers are
?

22/3,K/1 (Item 1 from file: 348)
DIALOG(R)File 348:EUROPEAN PATENTS

01033702
METHOD FOR THE TRANSMISSION OF SPEECH INACTIVITY WITH REDUCED POWER IN A TDMA SYSTEM
VERFAHREN ZUR UBERTRAGUNG VON SPRACHINAKTIVITAT SIGNALEN MIT REDUZIERTER LEISTUNG IN EINER TDMA ANORDNUNG
PROCEDE D'EMISSION A PUISSANCE REDUITE PENDANT L'ABSENCE DE PAROLE DANS UN SYSTEME AMRT
PATENT ASSIGNEE:
Telefonaktiebolaget L M Ericsson (Publ), (213764), 126 25 Stockholm, (SE), (Proprietor designated states: all)
INVENTOR:
BRUHN, Stefan, Fridskyddevagen 3, SE-19136 Sollentuna, (SE)
LEGAL REPRESENTATIVE:
Mohsler, Gabriele et al (84051), Ericsson Eurolab Deutschland GmbH, Research Department, Ericsson Allee 1, 52134 Herzogenrath, (DE)
PATENT (CC, No, Kind, Date): EP 1010267 A1 000621 (Basic)
EP 1010267 B1 020227
WO 9910995 990304
APPLICATION (CC, No, Date): EP 98947429 980813; WO 98EP5139 980813
PRIORITY (CC, No, Date): US 56444 P 970825; US 115632 980715
DESIGNATED STATES: DE; ES; FR; GB; IT
INTERNATIONAL PATENT CLASS: H04B-007/26; H04Q-007/30
NOTE:
No A-document published by EPO
LANGUAGE (Publication,Procedural,Application): English; English; English
FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS B	(English)	200209	1080
CLAIMS B	(German)	200209	936
CLAIMS B	(French)	200209	1346
SPEC B	(English)	200209	3859
Total word count - document A			0
Total word count - document B			7221
Total word count - documents A + B			7221

...CLAIMS frame rate higher than r or equal to r for the transmission of the channel **error** protected **speech** inactivity **frames** (FIRST SID, SID UPDATE);
repeatedly sending the **encoded** **speech** inactivity **frames** (SID UPDATE) during the periods of inactive **speech** in case the transmission **frame** rate is equal to r, and
reducing the transmission power by an amount such that...

22/3,K/2 (Item 1 from file: 349)
DIALOG(R)File 349:PCT FULLTEXT

00830860
PARTIAL REDUNDANCY ENCODING OF SPEECH
CODAGE PARTIEL, AVEC REDONDANCE, DE PAROLE
Patent Applicant/Assignee:
TELEFONAKTIEBOLAGET LM ERICSSON (publ), S-126 25 Stockholm, SE, SE (Residence), SE (Nationality)
Inventor(s):
EKUDDEN Erik, Fjarilsvagen 23, S-184 38 Akersberga, SE,

SJOBERG Johan, Karlbergsvagen 62, 1 tr, S-113 35 Stockholm, SE,
Legal Representative:
GULLSTRAND Malin (agent), Ericsson Radio Systems AB, Patent Unit Research, S-164 80 Stockholm, SE,
Patent and Priority Information (Country, Number, Date):
Patent: WO 200163774 A1 20010830 (WO 0163774)
Application: WO 2001SE394 20010222 (PCT/WO SE0100394)
Priority Application: US 2000183846 20000222; US 2001789691 20010220
Designated States: AE AG AL AM AT AU AZ BA BB BG BR BY BZ CA CH CN CR CU CZ DE DK DM DZ EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ LC LK LR LS LT LU LV MA MD MG MK MN MW MX MZ NO NZ PL PT RO RU SD SE SG SI SK SL TJ TM TR TT TZ UA UG UZ VN YU ZA ZW
(EP) AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE TR
(OA) BF BJ CF CG CI CM GA GN GW ML MR NE SN TD TG
(AP) GH GM KE LS MW MZ SD SL SZ TZ UG ZW
(EA) AM AZ BY KG KZ MD RU TJ TM
Publication Language: English
Filing Language: English
Fulltext Word Count: 4302

Fulltext Availability:
Claims

Claim
... a
telecommunications network, said encoded speech data being divided into a plurality of respective encoded **speech** **frames** , the method comprising:
sorting at least one of said plurality of **speech** **frames** having respective encoded **speech** data therein, said respective encoded speech data having a
predeten-nined **error** sensitivity characteristic associated therewith;
generating partial **redundant** data corresponding to said sorted **encoded**

speech data within said at least one **speech** **frame** ; and
1 0 transmitting a data packet containing said sorted encoded speech data and said...

...link, said codec
comprising:
sorting means for sorting at least one of a plurality of **speech** **frames**

having encoded **speech** data therein, said respective encoded speech data having a
predetermined **error** sensitivity characteristic associated therewith;
and
generating means for generating partial **redundant** data corresponding to said sorted **encoded** **speech** data within said at least one **speech** **frame** .

15 The codec according to claim 14, further comprising:
SUBSTITUTE SHEET (RULE 26)
transmitting means...

22/3,K/3 (Item 2 from file: 349)
DIALOG(R)File 349:PCT FULLTEXT

00553139 **Image available**
RATE DETECTION IN RADIO COMMUNICATION SYSTEMS

DETECTION DE DEBIT DANS DES SYSTEMES DE RADIOCOMMUNICATION
Patent Applicant/Assignee:
ERICSSON INC,
Inventor(s):
RAMESH Rajaram,
BOTTOMLEY Gregory E,
Patent and Priority Information (Country, Number, Date):
Patent: WO 200016512 A1 20000323 (WO 0016512)
Application: WO 99US17564 19990803 (PCT/WO US9917564)
Priority Application: US 98152063 19980911
Designated States: AE AL AM AT AU AZ BA BB BG BR BY CA CH CN CU CZ DE DK EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ LC LK LR LS LT LU LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK SL TJ TM TR TT UA UG UZ VN YU ZA ZW GH GM KE LS MW SD SL SZ UG ZW AM AZ BY KG KZ MD RU TJ TM AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE BF BJ CF CG CI CM GA GN GW ML MR NE SN TD TG
Publication Language: English
Fulltext Word Count: 10108
Fulltext Availability:
Detailed Description

Detailed Description
... baseband system model according to the third embodiment is shown in FIG. 6. During one **frame** period, the **speech** encoder can be modeled by a variable rate information bit source 600, which produces Nb...

...k); k = 0, Nb(M) - 1 1 according to the frame's rate m. The **error** detection **encoding** , convolutional **encoding** , **repeat** coding, interleaving, scrambling, and power control bit puncturing are represented by a block encoder 610...

22/3,K/4 (Item 3 from file: 349)
DIALOG(R)File 349:PCT FULLTEXT

00356414 **Image available**
ERROR PROTECTION IN DYNAMIC BIT ALLOCATION SUB-BAND CODING
PROTECTION CONTRE LES ERREURS DANS LE CODAGE DE SOUS-BANDE A REPARTITION DYNAMIQUE DES BITS
Patent Applicant/Assignee:
ERICSSON INC,
Inventor(s):
ZINSER Richard L,
KOCH Steven R,
Patent and Priority Information (Country, Number, Date):
Patent: WO 9638928 A1 19961205
Application: WO 96US8048 19960530 (PCT/WO US9608048)
Priority Application: US 95460000 19950602
Designated States: AL AM AT AU AZ BB BG BR BY CA CH CN CZ DE DK EE ES FI GB GE HU IS JP KE KG KP KR KZ LK LR LS LT LU LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK TJ TM TR TT UA UG UZ VN KE LS MW SD SZ UG AM AZ BY KG KZ MD RU TJ TM AT BE CH DE DK ES FI FR GB GR IE IT LU MC NL PT SE BF BJ CF CG CI CM GA GN ML MR NE SN TD TG
Publication Language: English
Fulltext Word Count: 5028
Fulltext Availability:
Detailed Description

Detailed Description

... of the binary code
assigned to each band's energy level are specially protected using **redundant** bits in the ·**encoding** process.

During decoding, **errors** in the protected bits are connected using a majority vote correction algoritlun. In addition to...confidence scores
exceeds a threshold, a muting analysis is performed. When a predetermined number of **frames** of . **speech** have energy value confidence scores greater than the threshold, a muting operation is performed for the energies in all of the sub-bands. If after muting, a predetermined number of **frames** of **speech** have energy value 25 confidence scores less than the threshold, the muting operation is disabled...

22/3,K/5 (Item 4 from file: 349)
DIALOG(R)File 349:PCT FULLTEXT

00240315
INCREASED SPEECH INTERLEAVE WITH REDUCED DELAY
SYSTEME D'ENTRELACEMENT ACCRU DE SIGNAUX VOCAUX, A TEMPS DE TRANSMISSION REDUIT
Patent Applicant/Assignee:
MOTOROLA INC,
Inventor(s):
SPEAR Stephen Lee,
Patent and Priority Information (Country, Number, Date):
Patent: WO 9314584 A1 19930722
Application: WO 93US242 19930112 (PCT/WO US9300242)
Priority Application: US 92357 19920117
Designated States: AU CA DE FI GB SE AT BE CH DE DK ES FR GB GR IE IT LU MC NL PT SE
Publication Language: English
Fulltext Word Count: 2811

Fulltext Availability:
Detailed Description

Detailed Description
... full
rate speech coding technique where each (20ms) block of (continuous) speech is digitized and **error** **encoded** for radio transmission in a **repeating** TDM time slot over eight frames - a so-called interleave depth of eight. In other words, it takes eight frames to recover all segment of the original 20ms block of **speech** . The transmission bit rate and **frame** length is such that the delay between the speech being spoken and being received and...

...the GSIVI system, it is
envisioned that while the full-rate user will use every **frame** for **speech** transmission, the half@rate user will use every 1 5 other, alternate frame for its...with error protection).

The Interleaver controls the transmitter (Tx) to interleave the 20 ms of **encoded** speech (along with **redundant** **error** protecting information from the CODEC) into a recurrent time slot over eight Time Division Multiplexed (TDM) **frames** . The encoded and

27/3,K/1   (Item 1 from file: 348)
DIALOG(R)File 348:EUROPEAN PATENTS

01271136
VOICE ENCODER AND VOICE ENCODING METHOD
SPARCHKODIERER UND SPRACHKODIERUNGSVERFAHREN
VOCODEUR ET PROCEDE CORRESPONDANT
PATENT ASSIGNEE:
  MATSUSHITA ELECTRIC INDUSTRIAL CO., LTD., (216883), 1006, Oaza-Kadoma, Kadoma-shi, Osaka 571-8501, (JP), (Applicant designated States: all)
INVENTOR:
  YASUNAGA, Kazutoshi, 3-33-17-305, Sugao, Miyamae-ku, Kawasaki-shi, Kanagawa 216-0015, (JP)
  MORII, Toshiyuki, 2-3-7-501, Nijigaoka, Asao-ku, Kawasaki-shi, Kanagawa 215-0015, (JP)
LEGAL REPRESENTATIVE:
  Grunecker, Kinkeldey, Stockmair & Schwanhausser Anwaltssozietat (100721), Maximilianstrasse 58, 80538 Munchen, (DE)
PATENT (CC, No, Kind, Date): EP 1132892 A1 010912 (Basic)
                             WO 200115144 010301
APPLICATION (CC, No, Date): EP 2000954908 000823; WO 2000JP5621 000823
PRIORITY (CC, No, Date): JP 99235050 990823; JP 99236728 990824; JP 99248363 990902
DESIGNATED STATES: AT; BE; CH; CY; DE; DK; ES; FI; FR; GB; GR; IE; IT; LI; LU; MC; NL; PT
EXTENDED DESIGNATED STATES: AL; LT; LV; MK; RO; SI
INTERNATIONAL PATENT CLASS: G10L-019/04; G10L-101:12
ABSTRACT WORD COUNT: 140

LANGUAGE (Publication,Procedural,Application): English; English; Japanese
FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS A	(English)	200137	2633
SPEC A	(English)	200137	14219
Total word count - document A			16852
Total word count - document B			0
Total word count - documents A + B			16852

...SPECIFICATION algebraic codebook, (5) dispersion pattern selected from among several arbitrarily prepared dispersion pattern candidates by **repeating** **encoding** and decoding of the **speech** signal and an subjective (listening) evaluation of the synthesized **speech** so that synthesized **speech** of high quality can be output and (6) dispersion pattern created based on phonological knowledge...algebraic codebook, (5) dispersion pattern selected from among several arbitrarily prepared dispersion pattern candidates by **repeating** **encoding** and decoding of the **speech** signal and subjective(listening) evaluation of the synthesized **speech** so that synthesized **speech** of high quality can be output and (6) dispersion pattern created based on phonological knowledge ...
?

29/3,K/1 (Item 1 from file: 348)
DIALOG(R)File 348:EUROPEAN PATENTS

00260543
System for continuous speech recognition.
System zur kontinuierlichen Spracherkennung.
Systeme de reconnaissance de la parole continue.
PATENT ASSIGNEE:
KABUSHIKI KAISHA TOSHIBA, (213130), 72, Horikawa-cho Saiwai-ku, Kawasaki-shi Kanagawa-ken 210, (JP), (applicant designated states: DE;FR;GB)
INVENTOR:
Nitta, Tsuneo c/o Patent Division, Kabushiki Kaisha Toshiba 1-1 Shibaura 1-chome, Minato-ku Tokyo 105, (JP)
Uehara, Kensuke c/o Patent Division, Kabushiki Kaisha Toshiba 1-1 Shibaura 1-chome, Minato-ku Tokyo 105, (JP)
Watanabe, Sadakazu c/o Patent Division, Kabushiki Kaisha Toshiba 1-1 Shibaura 1-chome, Minato-ku Tokyo 105, (JP)
LEGAL REPRESENTATIVE:
Henkel, Feiler, Hanzel & Partner (100401), Mohlstrasse 37, W-8000 Munchen 80, (DE)
PATENT (CC, No, Kind, Date): EP 265692 A1 880504 (Basic)
EP 265692 B1 920408
APPLICATION (CC, No, Date): EP 87114236 870929;
PRIORITY (CC, No, Date): JP 86227961 860929
DESIGNATED STATES: DE; FR; GB
INTERNATIONAL PATENT CLASS: G10L-005/06;
ABSTRACT WORD COUNT: 144

LANGUAGE (Publication,Procedural,Application): English; English; English
FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS B	(English)	EPBBF1	1284
CLAIMS B	(German)	EPBBF1	603
CLAIMS B	(French)	EPBBF1	897
SPEC B	(English)	EPBBF1	6133
Total word count - document A			0
Total word count - document B			8917
Total word count - documents A + B			8917

...CLAIMS value, and a means of comparing the total value with the standard value.

12. A **speech** recognition method comprising the steps of:
- extracting prescribed feature parameters from input signals for continuous **speech** ;
- continuously matching of the extracted feature parameters with a **voice** dictionary compiled of phonetic segment units having prescribed phonetic meanings and for obtaining similarities on the phonetic segment units;
- extracting a **sequence** comprising a predetermined **number** of phonetic segment likelihoods based on the similarities; and
- continuously combining the results of word...

...compiled phonetic segment units include continuant segments having a vowel steady part and a fricative **consonant** , **consonant** segments having transient parts to vowels, boundary segments expressing the boundary between a vowel and...

29/3,K/2 (Item 1 from file: 349)
DIALOG(R)File 349:PCT FULLTEXT

00557682 **Image available**
PHONOLOGICAL AWARENESS, PHONOLOGICAL PROCESSING, AND READING SKILL TRAINING SYSTEM AND METHOD
SENSIBILISATION PHONOLOGIQUE, TRAITEMENT PHONOLOGIQUE ET SYSTEME ET PROCEDE D'APPRENTISSAGE DE LA LECTURE
Patent Applicant/Assignee:
  COGNITIVE CONCEPTS INC, Suite 300, 990 Grove Street, Evanston, IL 60201, US, US (Residence), US (Nationality)
Inventor(s):
  WASOWICZ Janet M, 207 Hamilton Street, Evanston, IL 60202, US,
Legal Representative:
  LOHSE Timothy W (agent), Gray Cary Ware & Freidenrich LLP, Attn: Patent Dept., 400 Hamilton Avenue, Palo Alto, CA 94301-1825, US,
Patent and Priority Information (Country, Number, Date):
  Patent: WO 200021055 A1 20000413 (WO 0021055)
  Application: WO 99US23518 19991007 (PCT/WO US9923518)
  Priority Application: US 98103354 19981007; US 99414393 19991006
Designated States: AL AM AT AU AZ BA BB BG BR BY CA CH CN CU CZ DE DK EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ LC LK LR LS LT LU LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK SL TJ TM TR TT UA UG UZ VN YU
  (EP) AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE
  (OA) BF BJ CF CG CI CM GA GN GW ML MR NE SN TD TG
  (AP) GH GM KE LS MW SD SL SZ TZ UG ZW
  (EA) AM AZ BY KG KZ MD RU TJ TM
Publication Language: English
Filing Language: English
Fulltext Word Count: 19528

Fulltext Availability:
  Detailed Description

Detailed Description
... between sound units). The complexity of the structure of the sound unit refers to the **number** and **sequence** of **consonants** and vowels. In this module, the number of **consonants** and vowels for the entire word is not changed, but instead for the onset only. For example, the module may preferably begin with a very simple **sound** structure of C ("s" for example), proceed to CC ("st" for example) and then finally...

29/3,K/3 (Item 2 from file: 349)
DIALOG(R)File 349:PCT FULLTEXT

00533665 **Image available**
SPEECH CODING APPARATUS AND SPEECH DECODING APPARATUS
DISPOSITIF DE CODAGE ET DE DECODAGE DE LA PAROLE
Patent Applicant/Assignee:
  MATSUSHITA ELECTRIC INDUSTRIAL CO LTD,
  MORII Toshiyuki,
  YASUNAGA Kazutoshi,
Inventor(s):
  MORII Toshiyuki,
  YASUNAGA Kazutoshi,
Patent and Priority Information (Country, Number, Date):

Patent: WO 9965017 A1 19991216
Application: WO 99JP3064 19990608 (PCT/WO JP9903064)
Priority Application: JP 98160119 19980609; JP 98258271 19980911
Designated States: CA CN JP KR US AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE
Publication Language: English
Fulltext Word Count: 10568

Fulltext Availability:
Detailed Description

Detailed Description
... other hand, an application of excitation with a large number of pulses such as random **number** **sequence** to coding at lower bit rates introduces a phenomenon that **sound** qualities greatly deteriorate mainly on voiced speeches. In order to improve the deterioration, a method...

...the complicated processing and sometimes generates an allophone caused by a judgement error on a **speech** signal.

As described above, there has been no algebraic codebook which matches
?

32/3,K/1 (Item 1 from file: 348)
DIALOG(R)File 348:EUROPEAN PATENTS

00475685
Method of speech recognition
Verfahren zur Spracherkennung
Procede de reconnaissance de parole
PATENT ASSIGNEE:
  MATSUSHITA ELECTRIC INDUSTRIAL CO., LTD., (216883), 1006, Oaza Kadoma, Kadoma-shi, Osaka-fu, 571, (JP), (applicant designated states: DE;FR;GB)
INVENTOR:
  Hoshimi, Masakatsu, 5-10-20-304, Sagamioono, Sagamihara-shi, Kanagawa-ken , (JP)
  Miyata, Maki, Shuwa Rejidensu 203, 5-8-7, Ogikubo, Suginami-ku, Tokyo, (JP)
  Hiraoka, Shoji, 7-6-5, Tsuchihashi, Miyamae-ku, Kawasaki, (JP)
  Niyada, Katsuyuki, 8-1-406, Minamidai 2-chome, Sagamihara-shi, Kanagawa-ken, (JP)
LEGAL REPRESENTATIVE:
  Pellmann, Hans-Bernd, Dipl.-Ing. (9227), Patentanwaltsburo Tiedtke-Buhling-Kinne & Partner Bavariaring 4, 80336 Munchen, (DE)
PATENT (CC, No, Kind, Date): EP 492470 A2 920701 (Basic)
                             EP 492470 A3 930512
                             EP 492470 B1 971015
APPLICATION (CC, No, Date): EP 91121856 911219;
PRIORITY (CC, No, Date): JP 90404866 901221; JP 917477 910125; JP 9158796 910322; JP 91170908 910711; JP 91234388 910913
DESIGNATED STATES: DE; FR; GB
INTERNATIONAL PATENT CLASS: G10L-005/06;
ABSTRACT WORD COUNT: 223

LANGUAGE (Publication,Procedural,Application): English; English; English
FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS B	(English)	9710W2	3478
CLAIMS B	(German)	9710W2	2873
CLAIMS B	(French)	9710W2	4401
SPEC B	(English)	9710W2	18185
Total word count - document A			0
Total word count - document B			28937
Total word count - documents A + B			28937

...SPECIFICATION similarity interval has a low reliability, a wrong recognition tends to be caused if DP **matching** is done with **equal** weights being used over the whole of the input- **speech** interval. The phoneme standard patterns for calculating the similarities are generated for the vowel intervals and the **consonant** intervals. Therefore, during a silent interval, the similarities tend to be small with respect to...

32/3,K/2 (Item 1 from file: 349)
DIALOG(R)File 349:PCT FULLTEXT

00783384 **Image available**
SYSTEM AND METHOD FOR CLASSIFICATION OF SOUND SOURCES
SYSTEME ET PROCEDE DE CLASSIFICATION DE SOURCES SONORES
Patent Applicant/Assignee:

WAVEMAKERS RESEARCH INC, 3328 West 2nd Avenue, Vancouver, British Columbia V6R 1J1, CA, CA (Residence), CA (Nationality), (For all designated states except: US)
Patent Applicant/Inventor:
ZAKARAUSKAS Pierre, 1723 Kennington Road, Encinitas, CA 92024, US, US (Residence), CA (Nationality), (Designated only for: US)
Legal Representative:
J D HARRIMAN II (agent), COUDERT BROTHERSn P.C., 333 South Hope Street, 23RD Floor, Los Angeles, CA 90071, US,
Patent and Priority Information (Country, Number, Date):
Patent: WO 200116937 A1 20010308 (WO 0116937)
Application: WO 2000US23754 20000829 (PCT/WO US0023754)
Priority Application: US 99385975 19990830
Parent Application/Grant:
Related by Continuation to: US 99385975 19990830 (CON)
Designated States: AE AG AL AM AT AU AZ BA BB BG BR BY BZ CA CH CN CR CU CZ DE DK DM DZ EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ LC LK LR LS LT LU LV MA MD MG MK MN MW MX MZ NO NZ PL PT RO RU SD SE SG SI SK SL TJ TM TR TT TZ UA UG US UZ VN YU ZA ZW
(EP) AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE
(OA) BF BJ CF CG CI CM GA GN GW ML MR NE SN TD TG
(AP) GH GM KE LS MW MZ SD SL SZ TZ UG ZW
(EA) AM AZ BY KG KZ MD RU TJ TM
Publication Language: English
Filing Language: English
Fulltext Word Count: 6023

Fulltext Availability:
Claims

Claim
amendments.
SYSTEM AND METHOD FOR
CLASSIFICATION OF **SOUND** SOURCES
TECHNICAL FIELD
This invention relates to systems and methods for automatic classification of acoustic (**sound**) sources, including text-independent speaker identification.
BACKGROUND
There are several fields of research studying acoustic...

...acoustic signal classification, with some overlap between them. At present, the
main applications for automatic **sound** source classification are: speaker
verification; speaker identification; passive sonar classification; and machine noise monitoring or...

...the way that keyword was said by the putative speaker with training samples of the **same** keywords. If the **match** is poor, the speaker is rejected or denied service (e.g., computer or
premise access...

...the use of particular keywords.
Passive sonar classification involves identifying a vessel according to the **sound** it radiates underwater. Machine noise monitoring and diagnostics -Iinvolves determining the state of a piece of machinery through the **sound** it makes.
In all of the above applications, a model of each **sound** source is first obtained by training a system with a set of example sounds from each source.

A test sample is hen compared to the stored models to determine a **sound** source category for the test sample. Known methods reuire relatively long training times and testing...

...prior techniques.

SUMMARY

The invention includes a method, apparatus, and computer program to classify a **sound** source. The invention matches the acoustic input to a number of signal models, one per source class, and produces a score for each signal model. The **sound** source is declared to be of the same class as that of the model with...

...the use of a signal model augmented by leaming. The input signal may represent human **speech** , in which case the goal would be to identify the speaker in a text-independent...

...the following advantages: It is able to classify an acoustic signal source: independently of the **sound** the source happens to be emitting at the time of sampling; independently of **sound** levels; and even when some portions of the spectra of the acoustic signal are masked...26, or used by the system to customize its response to the identity of the **sound** source, or used to actuate external equipment (e.g., lock mechanisms in an access control...

...filter to apply to the data since in many cases of interest (e.g., human **voice** , music, bird singing, engine and machinery), the signal has a harmonic structure. A preferred embodiment...

...can be forced at any desired time or event (for example, if a period of **speech** is followed by a significant period of silence), and the best fitting class returned along...input signal and therefore for which no further comparison is necessary. For example, the human **voice** is characterized by the presence of a set of harmonics between 0. I and about...a typical embodiment, 8-20 scores are accumulated, each corresponding to a buffer of voiced **speech** (as opposed to unvoiced **speech** - **consonants** - since the buffers without voiced **speech** do not contain as much information as to the identity of the speaker. The classification...

?

35/3,K/1 (Item 1 from file: 348)
DIALOG(R)File 348:EUROPEAN PATENTS

00313502
Code excited linear predictive vocoder and method of operation.
Linearer Pradiktionsvocoder mit Code-Anregung.
Vocoder a prediction lineaire excite par codes.
PATENT ASSIGNEE:
AMERICAN TELEPHONE AND TELEGRAPH COMPANY, (589370), 550 Madison Avenue, New York, NY 10022, (US), (applicant designated states: AT;BE;DE;FR;GB;IT;NL;SE)
INVENTOR:
Ketchum, Richard Harry, 1754C Plymouth Court, Wheaton Illinois 60187, (US)
Kleijn, Willem Bastiaan, 238 North Van Nortwick, Batavia Illinois 60510, (US)
Krasinski, Daniel John, 1407 Fairway Drive, Glendale Heights Illinois 60139, (US)
LEGAL REPRESENTATIVE:
Watts, Christopher Malcolm Kelway, Dr. et al (37392), AT&T (UK) LTD. AT&T Intellectual Property Division 5 Mornington Road, Woodford Green Essex IG8 OTU, (GB)
PATENT (CC, No, Kind, Date): EP 296764 A1 881228 (Basic)
EP 296764 B1 920909
APPLICATION (CC, No, Date): EP 88305526 880617;
PRIORITY (CC, No, Date): US 67650 870626
DESIGNATED STATES: AT; BE; DE; FR; GB; IT; NL; SE
INTERNATIONAL PATENT CLASS: G10L-009/14;
ABSTRACT WORD COUNT: 160

LANGUAGE (Publication,Procedural,Application): English; English; English
FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS B	(English)	EPBBF1	1295
CLAIMS B	(German)	EPBBF1	838
CLAIMS B	(French)	EPBBF1	1120
SPEC B	(English)	EPBBF1	5349
Total word count - document A			0
Total word count - document B			8602
Total word count - documents A + B			8602

...CLAIMS in that
said communicating step further communicates the location of the selected other candidate excitation **frame** in said stochastic code book **for** reproduction of said **speech** for said present **speech** **frame** .
5. An apparatus for **encoding** **speech** **based** on **determining** sets of filter coefficients and corresponding excitation **frames** , said **speech** comprising **frames** each having a **plurality** of samples, comprising
means (101) for determining a set of filter coefficients of a filter...

35/3,K/2 (Item 1 from file: 349)
DIALOG(R)File 349:PCT FULLTEXT

00799836 **Image available**

DIGITAL AUDIO DECODER AND RELATED METHODS
DECODEUR AUDIO-NUMERIQUE ET PROCEDES CONNEXES

Patent Applicant/Assignee:
SONY ELECTRONICS INC, 1 Sony Drive, Park Ridge, NJ 07656, US, US (Residence), US (Nationality)

Inventor(s):
CHEN Hua, 1043 Danbury Drive, San Jose, CA 95129, US,
TSUKAGOSHI Ikuo, 718 Old San Francisco Road, Sunnyvale, CA 94086, US,
MEHTA Milan, 43555 Grimmer Blvd. #K289, Fremont, CA 94538, US,

Legal Representative:
GALLENSON Mavis (et al) (agent), Ladas & Parry, 5670 Wilshire Blvd., Suite 2100, Los Angeles, CA 90036, US,

Patent and Priority Information (Country, Number, Date):
Patent: WO 200133405 A1 20010510 (WO 0133405)
Application: WO 2000US41450 20001019 (PCT/WO US0041450)
Priority Application: US 99422134 19991020

Designated States: AE AG AL AM AT AU AZ BA BB BG BR BY BZ CA CH CN CR CU CZ DE DK DM DZ EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ LC LK LR LS LT LU LV MA MD MG MK MN MW MX MZ NO NZ PL PT RO RU SD SE SG SI SK SL TJ TM TR TT TZ UA UG UZ VN YU ZA ZW
(EP) AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE
(OA) BF BJ CF CG CI CM GA GN GW ML MR NE SN TD TG
(AP) GH GM KE LS MW MZ SD SL SZ TZ UG ZW
(EA) AM AZ BY KG KZ MD RU TJ TM

Publication Language: English
Filing Language: English
Fulltext Word Count: 9631

Fulltext Availability:
Claims

Claim
... 8 further comprising the steps of:
c) provided an error is detected in a next **encoded** audio frame immediately following said current **encoded** audio frame, repeating said current **encoded** audio **frame** in lieu of said next **encoded** audio **frame** ; said step c) comprising the steps of:
c1) obtaining decoded data of said current **encoded** audio **frame** ;
c2) generating a second repeated audio **frame** by replicating said decoded data of said current **encoded** audio **frame** for use in lieu of said next
encoded audio **frame** ;
c3) modifying said second repeated audio **frame** by adding delay information of a last block of said current **encoded** audio **frame** with pulse code modulated (PCM) data of a first block of said second repeated audio frame to generate new decoded data for said first block of said second repeated **audio**
frame; and
b4) sending said second repeated audio frame to an audio output buffer for...
?

37/3,K/1 (Item 1 from file: 348)
DIALOG(R)File 348:EUROPEAN PATENTS

01624231
Method and apparatus for performing reduced rate variable rate vocoding
Verfahren und Vorrichtung zur Sprachkodierung mit reduzierter, variabler Bit-Rate
Procede et dispositif de codage de la parole a bas debit reduit et variable
PATENT ASSIGNEE:
QUALCOMM Incorporated, (910896), 5775 Morehouse Drive, San Diego, California 92121, (US), (Applicant designated States: all)
INVENTOR:
Dejaco, Andrew P. c/o Qualcomm Incorporated, 5775 Morehouse Drive, San Diego, CA 92121, (US)
LEGAL REPRESENTATIVE:
Wagner, Karl H., Dipl.-Ing. et al (12567), Wagner & Geyer, Patentanwalte, Gewurzmuhlstrasse 5, 80538 Munchen, (DE)
PATENT (CC, No, Kind, Date): EP 1339044 A2 030827 (Basic)
APPLICATION (CC, No, Date): EP 2003005273 950801;
PRIORITY (CC, No, Date): US 286842 940805
DESIGNATED STATES: AT; BE; CH; DE; DK; ES; FR; GB; GR; IE; IT; LI; LU; MC; NL; PT; SE
EXTENDED DESIGNATED STATES: LT; LV; SI
RELATED PARENT NUMBER(S) - PN (AN):
EP 722603 (EP 95928266)
INTERNATIONAL PATENT CLASS: G10L-019/12
ABSTRACT WORD COUNT: 106
NOTE:
Figure number on first page: NONE

LANGUAGE (Publication,Procedural,Application): English; English; English
FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS A	(English)	200335	1399
SPEC A	(English)	200335	7637
Total word count - document A			9036
Total word count - document B			0
Total word count - documents A + B			9036

...SPECIFICATION a normalized autocorrelation measurement indicative of the periodicity in the input speech, a target matching **signal** to **noise ratio** measurement indicative of match between an **encoded frame** of **speech** and an input **frame** of **speech** , and a prediction gain differential measurement indicative of the **frame** to **frame** stability of a set of formant parameters in said **encoded speech frame** and wherein when normalized autocorrelation measurement exceeds a predetermined **first** threshold, said prediction gain differential exceeds a **second** predetermined threshold and said normalized autocorrelation function exceeds a predetermined third threshold said step of...

37/3,K/2 (Item 1 from file: 349)
DIALOG(R)File 349:PCT FULLTEXT

00322138 **Image available**
METHOD AND APPARATUS FOR PERFORMING REDUCED RATE VARIABLE RATE VOCODING
PROCEDE ET APPAREIL PERMETTANT D'EFFECTUER LE CODAGE DE LA VOIX A VITESSE

VARIABLE, VITESSE REDUITE

Patent Applicant/Assignee:
QUALCOMM INCORPORATED,
Inventor(s):
DEJACO Andrew P,
Patent and Priority Information (Country, Number, Date):
Patent: WO 9604646 A1 19960215
Application: WO 95US9780 19950801 (PCT/WO US9509780)
Priority Application: US 94842 19940805
Designated States: AM AT AU BB BG BR BY CA CH CN CZ DE DK EE ES FI GB GE HU IS JP KE KG KP KR KZ LK LR LT LU LV MD MG MN MW MX NO NZ PL PT RO RU SD SE SG SI SK TJ TM TT UA UG UZ VN KE MW SD SZ UG AT BE CH DE DK ES FR GB GR IE IT LU MC NL PT SE BF BJ CF CG CI CM GA GN ML MR NE SN TD TG
Publication Language: English
Fulltext Word Count: 7950

Fulltext Availability:
Claims

Claim
... a normalized autocorrelation measurement indicative of the periodicity in the input speech, a target matching **signal** to **noise ratio** measurement indicative of match between an **encoded** **frame** of **speech** and an input **frame** of **speech** , and a prediction gain differential measurement indicative of the **frame** to **frame** stability of a set of formant parameters in said **encoded** **speech** **frame** and wherein when normalized autocorrelation measurement exceeds a predetermined **first** threshold, said prediction gain differential exceeds a **second** predetermined threshold and said normalized autocorrelation function exceeds a predetermined third threshold said rate determination...a normalized autocorrelation measurement indicative of the periodicity in the input speech, a target matching **signal** to **noise ratio** measurement indicative of match between an **encoded** **frame** of **speech** and an input **frame** of **speech** , and a prediction gain differential measurement indicative of the **frame** to **frame** stability of a set of formant parameters in said **encoded** **speech** **frame** and wherein when normalized autocorrelation measurement exceeds a predetermined **first** threshold, said prediction gain differential exceeds a **second** predetermined threshold and said normalized autocorrelation function exceeds a predetermined third threshold said rate determination...a normalized autocorrelation measurement indicative of the periodicity in the input speech, a target matching **signal** to **noise ratio** measurement indicative of match between an **encoded** **frame** of **speech** and an input **frame** of **speech** , and a prediction gain differential measurement indicative of the **frame** to **frame** stability of a set of formant parameters in said **encoded** **speech** **frame** and wherein when normalized

autocorrelation
measurement exceeds a predetermined **first** threshold, said prediction gain differential exceeds a **second** predetermined threshold and said normalized autocorrelation function exceeds a predetermined third threshold said step of...

?

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File 344:Chinese Patents Abs Aug 1985-2004/Mar
         (c) 2004 European Patent Office
File 347:JAPIO Nov 1976-2003/Nov(Updated 040308)
         (c) 2004 JPO & JAPIO
File 350:Derwent WPIX 1963-2004/UD,UM &UP=200417
         (c) 2004  Thomson Derwent

Set     Items   Description
S1      428875  VOICE OR AUDIO OR SOUND OR SPEECH
S2      938470  FRAME?
S3        8782  (INTERPOLAT? OR ENCOD?) AND (REPEAT? OR ITERATIV? OR REDUN-
             DANT? OR REITERA?)
S4        1825  S3 AND (RECOVER? OR ERROR?)
S5       12291  SIGNAL(3N)NOISE()RATIO
S6        1830  VOIP OR VOICE(3N)INTERNET
S7        2590  S2 AND (FIRST OR INITIAL?) AND (SECOND OR SUBSEQUENT?) AND
             ENCOD?
S8        1662  PARAMETER? AND PACKET?
S9        2195  CONSONANT??
S10       9266  (INDEX OR SEQUENCE)(3N)NUMBER??
S11        126  S10 AND (MULTIPLEX? OR MULTI()PLEX?) AND (TRANSMIT? OR TRA-
             NSMIS? OR SEND OR SENDING OR SENDS)
S12      44681  (SAME OR CLOSE OR EQUAL OR EQUIVALENT OR APPROXIMAT? OR MA-
             XIMUM OR HIGHEST) AND MATCH?
S13      50972  IC=G10L?
S14         40  (S1 OR S6) AND S4 AND S2
S15         15  S14 AND S13
S16          2  S15 AND AD=20001128:20040315/PR
S17         13  S15 NOT S16
S18         13  IDPAT (sorted in duplicate/non-duplicate order)
S19         13  IDPAT (primary/non-duplicate records only)
S20          0  S7 AND S8 AND S11
S21          1  S7 AND S11
S22          1  S21 NOT S19
S23          3  S4 AND S11
S24          2  S23 NOT (S21 OR S19)
S25          0  S14 AND S5
S26          1  S2 AND S4 AND S5
S27          1  S26 NOT (S23 OR S21 OR S19)
S28          1  S7 AND S3 AND S5
S29          1  S28 NOT (S26 OR S23 OR S21 OR S19)
S30          0  S7 AND S8 AND S10
S31          0  S7 AND S8 AND S9
```

19/3,K/1 (Item 1 from file: 350)
DIALOG(R)File 350:Derwent WPIX

013985941 **Image available**
WPI Acc No: 2001-470155/200151
XRPX Acc No: N01-349142
Video and audio **information** encoding **and decoding procedure involves adding parity check bit to cyclic redundancy code test data which is added to information** encoded **per** frame
Patent Assignee: NEC IC MICROCOMPUTER SYSTEMS LTD (NIDE)
Number of Countries: 001 Number of Patents: 001
Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
JP 2001168731	A	20010622	JP 99348107	A	19991207	200151	B

Priority Applications (No Type Date): JP 99348107 A 19991207
Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
JP 2001168731	A		12	H03M-013/09	

Video and audio **information** encoding **and decoding procedure involves adding parity check bit to cyclic redundancy code test data which is added to information** encoded **per** frame
Abstract (Basic):
... Video and **audio** information are **encoded** per **frame** . Cyclic redundancy code (CRC) test data are added to each **encoded** **frame** . An odd number or even number parity check bit is added to cyclic redundancy code (CRC) test data to determined parity **error** .
... Transmission efficiency is improved, by decoding each **frame** of video and **audio** data irrespective of **error** in **frames** detected by cyclic redundancy code inspection method...
...Title Terms: **AUDIO ;**
International Patent Class (Additional): **G10L-019/00** ...

19/3,K/2 (Item 2 from file: 350)
DIALOG(R)File 350:Derwent WPIX

012944105 **Image available**
WPI Acc No: 2000-115958/200010
XRPX Acc No: N00-087807
Audio **coding method for electrical signal**
Patent Assignee: NOKIA MOBILE PHONES LTD (OYNO)
Inventor: YIN L
Number of Countries: 001 Number of Patents: 001
Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
US 6012025	A	20000104	US 9814712	A	19980128	200010	B

Priority Applications (No Type Date): US 9814712 A 19980128
Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
US 6012025	A		7	G10L-009/14	

Audio **coding method for electrical signal**

Abstract (Basic):
... A stream of spectral data values for each spectral component is generated by **repeating** the time **frame** transformation. A set of

prediction coefficients is computed using the covariances of the spectral data stream. The **error** between predicted and actual spectral data values is computed and recombined to predicted spectral values to reconstruct spectral values for producing coded **audio** signal.

... a) an **audio** decoding method for electrical signal...

...b) an **audio** **encoder ;**
(...

...c) an **audio** decoder...

...Has improved backward adaptive prediction algorithm that **encodes** relatively large number of frequencies of **audio** signal and calculates prediction coefficients from predetermined number of sample values. Enhances robustness of the backward adaptive prediction against channel **error** and numerical round-off **error** by performing bandwidth expansion after obtaining linear prediction coefficient. Offers labor saving in computing prediction...

...The figure shows the schematic diagram of the apparatus for **encoding** **audio** signal using backward adaptive prediction
Title Terms: **AUDIO ;**
International Patent Class (Main): **G10L-009/14**

19/3,K/3 (Item 3 from file: 350)
DIALOG(R)File 350:Derwent WPIX

011217211 **Image available**
WPI Acc No: 1997-195136/199718
XRPX Acc No: N97-161230
Code-excited linearly predicted speech **compression with low bit-rate - transmits** speech **over narrow bandwidth channel by A-D conversion of signal and breaking up of digitised** speech **into plural** frames
Patent Assignee: ROCKWELL INT CORP (ROCW)
Inventor: SU H
Number of Countries: 005 Number of Patents: 003
Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
EP 766231	A2	19970402	EP 96115299	A	19960924	199718	B
JP 9190198	A	19970722	JP 96254230	A	19960926	199739	
US 5664054	A	19970902	US 95536329	A	19950929	199741	

Priority Applications (No Type Date): US 95536329 A 19950929
Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
EP 766231	A2	E	14	G10L-009/14	
Designated States (Regional): DE FR GB					
JP 9190198	A		13	G10L-009/14	
US 5664054	A		12	G10L-009/04	

Code-excited linearly predicted speech **compression with low bit-rate...**

...**transmits** speech **over narrow bandwidth channel by A-D conversion of signal and breaking up of digitised** speech **into plural** frames

...Abstract (Basic): Conventionally, a code-excited linearly predictive **speech** **encoder** synthesises a pitch interval from a scaled innovation signal (44), e.g. random, and adding...

...The inventive concept improves this technique by replacing, e.g. at the start of a **sound** , the scaled innovation signal with a scaled 'spike' signal (112...

...signals are by definition designed to represent differences between adjacent pitch intervals within a given **sound** , rather than at its transient start...

...USE/ADVANTAGE - Improved transmission fidelity, as perceived by human ear, of intelligible synthesised **speech** , without requiring additional 'bit'-bandwidth, by exploiting differences between transient start of **speech** **sound** and content of progressive, steady-state **speech** .

...Abstract (Equivalent): A method for transmitting **speech** over a narrow bandwidth channel, the method comprising the steps of...

...a) converting **speech** from an analog auditory signal to an analog electronic signal...

...b) digitizing the electronic signal into digitized **speech** with an analog-to-digital converter...

...c) breaking the digitized **speech** into a plurality of **frames** ; (...

...d) selecting a next **frame** and applying it to...

...spike minimizer generating a spike gain code and a spike signal code which minimize an **error** between the scaled spike and the analysis filter output...

...the pitch minimizer, a pitch gain code and a pitch lag code which minimizes an **error** between...
...the innovation minimizer, an innovation gain code and an innovation signal code which minimize an **error** between...

...r) **repeating** steps (d) through (q) until the **speech** stops
...Title Terms: **SPEECH** ;
International Patent Class (Main): **G10L-009/04** ...

... **G10L-009/14**
International Patent Class (Additional): **G10L-009/18** ...

19/3,K/4 **(Item 4 from file: 350)**
DIALOG(R)File 350:Derwent WPIX

010135603 **Image available**
WPI Acc No: 1995-036854/199509
XRPX Acc No: N95-029015
Parameter management for speech **decoder - including synchronisation bits in** frames **and detecting incorrect synchronisation to inhibit use of** speech **decoding parameters until resynchronisation completed**
Patent Assignee: NOKIA TELECOM OY (OYNO)
Inventor: LEHTIMAKI M; VANSKA M; LEHTIMAEKI M; VAENSKAE M
Number of Countries: 022 Number of Patents: 012
Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
WO 9429982	A1	19941222	WO 94FI244	A	19940608	199509	B

FI 9302664	A	19941211	FI 932664	A	19930610	199510
AU 9468470	A	19950103	AU 9468470	A	19940608	199522
EP 654194	A1	19950524	EP 94917011	A	19940608	199525
			WO 94FI244	A	19940608	
FI 94817	B	19950714	FI 932664	A	19930610	199534
AU 670693	B	19960725	AU 9468470	A	19940608	199637
JP 8500231	W	19960109	WO 94FI244	A	19940608	199642
			JP 95501375	A	19940608	
CN 1110891	A	19951025	CN 94190369	A	19940608	199736
EP 654194	B1	19990113	EP 94917011	A	19940608	199907
			WO 94FI244	A	19940608	
DE 69415934	E	19990225	DE 615934	A	19940608	199914
			EP 94917011	A	19940608	
			WO 94FI244	A	19940608	
ES 2127928	T3	19990501	EP 94917011	A	19940608	199924
US 6208961	B1	20010327	WO 94FI244	A	19940608	200119
			WO 94FI244	A	19940608	
			US 95379584	A	19950320	
			US 97966234	A	19971107	

Priority Applications (No Type Date): FI 932664 A 19930610
Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
WO 9429982	A1	E	19	H04L-001/00	
Designated States (National): AU CN DE GB JP US					
Designated States (Regional): AT BE CH DE DK ES FR GB GR IE IT LU MC NL PT SE					
FI 9302664	A			H04L-025/49	
AU 9468470	A			H04L-001/00	Based on patent WO 9429982
EP 654194	A1	E	19	H04L-001/00	Based on patent WO 9429982
Designated States (Regional): AT BE CH DE DK ES FR GB GR IE IT LI LU MC NL PT SE					
FI 94817	B			H04L-025/49	Previous Publ. patent FI 9302664
AU 670693	B			H04L-001/00	Previous Publ. patent AU 9468470
					Based on patent WO 9429982
JP 8500231	W		18	H04L-001/00	Based on patent WO 9429982
CN 1110891	A			H04L-001/00	
EP 654194	B1	E		H04L-001/00	Based on patent WO 9429982
Designated States (Regional): AT BE CH DE DK ES FR GB GR IE IT LI LU MC NL PT SE					
DE 69415934	E			H04L-001/00	Based on patent EP 654194
					Based on patent WO 9429982
ES 2127928	T3			H04L-001/00	Based on patent EP 654194
US 6208961	B1			G10L-011/00	Cont of application WO 94FI244
					Cont of application US 95379584
					Based on patent WO 9429982

Parameter management for speech **decoder...**

...**including synchronisation bits in** frames **and detecting incorrect synchronisation to inhibit use of** speech **decoding parameters until resynchronisation completed**

...Abstract (Basic): The **speech** decoding method involves checking the synchronisation to control use of **speech** parameters. The **speech** decoder receives information **frames** containing **speech** **encoding** parameters. The first bits in each **frame** for a synchronisation word and a synchronisation check bit is **repeated** at regular intervals...

...A synchronisation unit (47) synchronises the **speech** decoder with the **frames** and extracts the **speech** **encoding** parameters which are held

in memories (43-47). If the synchronising unit detects **errors** in a **frame** , it prevents updating of the **speech** parameters until a correctly synchronised **frame** is found...

...ADVANTAGE - Avoids disturbances of the **speech** decoding due to incorrect **speech** **encoding** parameters being used...
...Title Terms: **SPEECH** ;
International Patent Class (Main): **G10L-011/00** ...

19/3,K/5 (Item 5 from file: 350)
DIALOG(R)File 350:Derwent WPIX

009932755 **Image available**
WPI Acc No: 1994-200466/199424
XRPX Acc No: N94-157624
Coding digital data with modulation error **control weighting and bit allocation - using data** encoding **to allow incorrectable bit** errors **to be detected without requiring further redundancy to be added to data stream**
Patent Assignee: DIGITAL VOICE SYSTEMS INC (DIGI-N)
Inventor: HARDWICK J C; LIM J S
Number of Countries: 044 Number of Patents: 011
Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
WO 9412932	A1	19940609	WO 93US11609	A	19931129	199424	B
AU 9456833	A	19940622	AU 9456833	A	19931129	199436	
EP 671032	A1	19950913	WO 93US11609	A	19931129	199541	
			EP 94902473	A	19931129		
US 5517511	A	19960514	US 92982937	A	19921130	199625	
EP 671032	A4	19960814	EP 94902473	A		199702	
US 5870405	A	19990209	US 92982937	A	19921130	199913	
			US 96610184	A	19960304		
EP 955586	A1	19991110	EP 94902473	A	19931129	199952	
			EP 99114399	A	19931129		
EP 671032	B1	20000308	WO 93US11609	A	19931129	200017	
			EP 94902473	A	19931129		
			EP 99114399	A	19931129		
DE 69328034	E	20000413	DE 628034	A	19931129	200025	
			WO 93US11609	A	19931129		
			EP 94902473	A	19931129		
EP 955586	B1	20020502	EP 94902473	A	19931129	200230	
			EP 99114399	A	19931129		
DE 69331886	E	20020606	DE 631886	A	19931129	200245	
			EP 99114399	A	19931129		

Priority Applications (No Type Date): US 92982937 A 19921130; US 96610184 A 19960304
Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
WO 9412932	A1	E	68	G06F-011/10	

Designated States (National): AT AU BB BG BR BY CA CH CZ DE DK ES FI GB HU JP KP KR KZ LK LU MG MN MW NL NO NZ PL PT RO RU SD SE SK UA US VN
Designated States (Regional): AT BE CH DE DK ES FR GB GR IE IT LU MC NL OA PT SE

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
AU 9456833	A				Based on patent WO 9412932
EP 671032	A1	E	68		Based on patent WO 9412932

Designated States (Regional): DE FR GB SE

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
US 5517511	A		32	G06F-011/08	

US 5870405	A	G06F-011/00	Div ex application US 92982937 Div ex patent US 5517511
EP 955586	A1 E		Div ex application EP 94902473 Div ex patent EP 671032
Designated States (Regional): DE FR GB SE			
EP 671032	B1 E	H03M-013/00	Related to application EP 99114399 Related to patent EP 955586 Based on patent WO 9412932
Designated States (Regional): DE FR GB SE			
DE 69328034	E	H03M-013/00	Based on patent EP 671032 Based on patent WO 9412932
EP 955586	B1 E	H03M-013/00	Div ex application EP 94902473 Div ex patent EP 671032
Designated States (Regional): DE FR GB SE			
DE 69331886	E	H03M-013/00	Based on patent EP 955586

Coding digital data with modulation error **control weighting and bit allocation...**
...**using data** encoding **to allow incorrectable bit** errors **to be detected without requiring further redundancy to be added to data stream**

...Abstract (Basic): The method involves **encoding** digital data by dividing digital data into one or more **frames** and further dividing each of the **frames** into several bit vectors. One or more of the vectors are coded with **error** control codes. A modulation key is generated from one or more of the bit vectors, and the modulation key is used to modulate one or more of the **encoded** bit vectors. The bit vectors are each **encoded** by a first type of **error** control code and second group of the bit vectors are **encoded** by a second type of **error** control code...

...The modulation key is generated from a high priority bit vector. The **frames** of digital area are generated by **encoding** a **speech** signal with a **speech** coder. The **frames** can be grouped into **frame** formats and the modulation key is generated from one bit vector determining the **frame** format used in the current **frame** .

...

...USE - For preserving quality of **speech** or other acoustic signals when transmitted over noisy channel
...Abstract (Equivalent): A method for **error** control coding of digital data, the method comprising the steps of...

... **encoding** said bit vectors with **error** control codes, to produce **encoded** bit vectors, including an **encoded** first bit vector...

...using said modulation key to modulate at least some of said **encoded** bit vectors
...Title Terms: **ERROR ;**
International Patent Class (Additional): **G10L-003/00** ...

... **G10L-003/02** ...

... **G10L-007/02** ...

... **G10L-009/02** ...

... **G10L-009/18** ...

... **G10L-019/00** ...

... **G10L-019/06**

19/3,K/6 (Item 6 from file: 350)
DIALOG(R)File 350:Derwent WPIX

008771445 **Image available**
WPI Acc No: 1991-275458/199138
XRPX Acc No: N91-210422

Search complexity reduction in analysis-by-synthesis coding - using limited search routine that only searches those areas which approximate to input signal

Patent Assignee: GTE LAB INC (SYLV)
Inventor: MAZOR B; VEENEMAN D E
Number of Countries: 007 Number of Patents: 007
Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
EP 446817	A	19910918	EP 91103623	A	19910308	199138	B
CA 2037475	A	19910916				199149	
US 5144671	A	19920901	US 90494071	A	19900315	199238	
EP 446817	A3	19920304	EP 91103623	A	19910308	199325	
EP 446817	B1	19970604	EP 91103623	A	19910308	199727	
DE 69126347	E	19970710	DE 626347	A	19910308	199733	
			EP 91103623	A	19910308		
CA 2037475	C	20010814	CA 2037475	A	19910304	200154	

Priority Applications (No Type Date): US 90494071 A 19900315
Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
EP 446817	A				
Designated States (Regional): BE DE FR GB IT					
US 5144671	A		9	G10L-005/00	
EP 446817	B1	E	9	G10L-009/14	
Designated States (Regional): BE DE FR GB IT					
DE 69126347	E			G10L-009/14	Based on patent EP 446817
CA 2037475	C	E		G10L-009/00	

...Abstract (Basic): The method, for **encoding** a **frame** of input **speech** using a tree-code excitation code book, comprises partitioning the **speech** **frame** into a predetermined number of sample segments of length equal to the length of each...

...of the tree-code is then searched, to find a codeword which represents the input **speech** signal, in such a way that at each stage of the tree-code only a...

...be searched, colouring the code letters of the branches with a colouring filter, minimising an **error** distance measurement between a synthetic signal identified by each search path and the sequence of sample segments, and saving the paths with the lowest distance measurements. The search method is **repeated** until only a single path, with the lowest relative distance, is saved...

...Abstract (Equivalent): A method of **encoding** a **frame** of input **speech** signal using a tree-code excitation codebook wherein each branch of said tree-code represents...

...said tree-code to find a code word achieving an optimal representation of said input **speech** signal, said search operating so that at each

stage of said tree-code only a...

...node, colouring the respective codeletters of said extended branches with a colouring filter, minimizing an **error** distance measurement between a synthetic signal defined by each path being currently searched and the **frame** of input **speech** up to the current stage; saving those limited number of paths having the lowest distance...

...the other currently searched paths; and the limited searching continues into the next stage by **repeating** the steps of path identification, codeletter colouring, **error** distance minimization by optimal scaling and path saving so that after reaching the last stage...

...having the lowest relative distance measurement represents the codeword achieving an optimal representation of said **frame** of input **speech** signal; characterised in that before performing said limited search of said tree code, the **speech** **frame** is partitioned into a predetermined number of sample segments of length equal to the length ...

...Abstract (Equivalent): gain calculation for each test path under consideration. The gain calculation occurs when minimising an **error** distance measurement between a synthetic signal defined by each test path being considered and the appropriate **speech** signal by optimising a scaling factor of the synthetic signal...

...The **encoding** method achieves a significant reduction in computational complexity with minimal loss of performance...

...USE - Of **encoding** **speech** .
International Patent Class (Main): **G10L-005/00** ...

... **G10L-009/00** ...

... **G10L-009/14**

19/3,K/7 (Item 7 from file: 350)
DIALOG(R)File 350:Derwent WPIX

008338902 **Image available**
WPI Acc No: 1990-225903/199030
XRPX Acc No: N90-175329
Low-delay code-excited linear predictive coder - iteratively **updating gain factor by logarithmic-based calculation that produces weighted average of previously calculated gains**
Patent Assignee: AMERICAN TELEPHONE & TELEGRAPH CO (AMTT); AT & T CORP (AMTT)
Inventor: CHEN J; CHEN J H
Number of Countries: 010 Number of Patents: 009
Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
EP 379296	A	19900725	EP 90300255	A	19900110	199030	B
AU 9047753	A	19900726				199038	
CA 2005115	A	19900717				199040	
JP 2231825	A	19900913				199043	
EP 379296	B1	19960508	EP 90300255	A	19900110	199623	
DE 69026843	E	19960613	DE 626843	A	19900110	199629	
			EP 90300255	A	19900110		
ES 2087124	T3	19960716	EP 90300255	A	19900110	199635	

CA 2005115	C	19970422	CA 2005115	A	19891211	199729
KR 161971	B1	19981201	KR 90457	A	19900116	200033

Priority Applications (No Type Date): US 89298451 A 19890117

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
EP 379296	A				
Designated States (Regional): DE ES FR GB IT NL					
EP 379296	B1	E	31	G10L-009/14	
Designated States (Regional): DE ES FR GB IT NL					
DE 69026843	E			G10L-009/14	Based on patent EP 379296
ES 2087124	T3			G10L-009/14	Based on patent EP 379296
CA 2005115	C			G10L-005/00	
KR 161971	B1			H04B-014/06	

... iteratively **updating gain factor by logarithmic-based calculation that produces weighted average of previously calculated gains**

...Abstract (Basic): The **encoding** method comprises the steps of grouping **speech** into **frames** of **speech** each having a number of samples with each **frame** representing a portion of the **speech** , and forming a target set of **speech** -related information in response to at least a portion of a **frame** of **speech** which is the current **frame** of **speech** . A set of synthesis filter coefficients are determined in response to at least a portion of a **frame** of **speech** and information representing a synthesis filter is calculated from the set of synthesis filter coefficients. An **error** value is **iteratively** calculated for each of a number of candidate sets of excitation information stored in a...

...to the filter information and each of the candidate sets and the target set of **speech** -related information. An adapted gain is by which each candidate set is multiplied is calculated prior, to calculating the respective **error** value. One of the candidate sets of excitation information is selected as producing the smallest **error** value. Information including information representing the location in the table of the selected one of the candidate sets of excitation information is communicated for reproduction of the **speech** for the current **frame** of **speech** .

...Abstract (Equivalent): A method of **encoding** **speech** for communication to a decoder for reproduction, comprising the steps of grouping said **speech** into **frames** of **speech** each having a plurality of samples with each **frame** representing a portion of said **speech** ; forming a target set of **speech** -related information in response to at least a portion of a **frame** of **speech** ; calculating information representing a synthesis filter from said set of synthesis filter coefficients; **iteratively** calculating an **error** value for each of a plurality of candidate sets of excitation information stored in a...

...to a filter information and each of said candidate sets and said target set of **speech** -related information, including calculating an adapted gain by which each candidate set is multiplied prior to calculating the respective **error** value; selecting one of said candidate sets of excitation information as producing the smallest **error** value; communicating information including information representing the location in the table of the selected one of said candidate sets of excitation information for reproduction of said **speech** for the current **frame** of **speech** , characterised in that the forming step includes forming a target set of **speech** -related information in

response to a portion of the current **frame** of **speech** , which portion is the current **speech** vector; the communicating step excludes communicating the set of synthesis filter coefficients, and the step...

...the set of synthesis filter coefficients includes determining them by linear predictive analysis from a **speech** vector representing at least a portion of a **frame** of simulated decoded **speech** , which vector occurred prior to the current **speech** vector...
...Title Terms: **ITERATIVE** ;
International Patent Class (Main): **G10L-005/00** ...

... **G10L-009/14**

19/3,K/8 (Item 8 from file: 350)
DIALOG(R)File 350:Derwent WPIX

007736112 **Image available**
WPI Acc No: 1989-001224/198901
XRPX Acc No: N89-000984
Code excited linear predictive vocoding for speech **- using virtual searching technique to improve performance during** speech **transitions**
Patent Assignee: AMERICAN TELEPHONE & TELEGRAPH CO (AMTT)
Inventor: KETCHUM R H; KLEIJN W B; KRASINSKI D J
Number of Countries: 012 Number of Patents: 007
Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
EP 296764	A	19881228	EP 88305526	A	19880617	198901	B
AU 8818378	A	19890105				198908	
US 4910781	A	19900320	US 8767650	A	19870626	199017	
EP 296764	B1	19920909	EP 88305526	A	19880617	199237	
DE 3874427	G	19921015	DE 3874427	A	19880617	199243	
			EP 88305526	A	19880617		
CA 1336455	C	19950725	CA 566911	A	19880516	199537	
KR 128066	B1	19980402	KR 887693	A	19880625	200009	

Priority Applications (No Type Date): US 8767650 A 19870626
Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
EP 296764	A	E	17		
Designated States (Regional): AT BE DE FR GB IT NL SE					
US 4910781	A		14		
EP 296764	B1	E	20	G10L-009/14	
Designated States (Regional): AT BE DE FR GB IT NL SE					
DE 3874427	G			G10L-009/14	Based on patent EP 296764
KR 128066	B1			G10L-003/00	
CA 1336455	C			G10L-009/14	

Code excited linear predictive vocoding for speech - ...

...**using virtual searching technique to improve p**erformance during speech **transitions**

...Abstract (Basic): The **speech** comprises **frames** each represented by a **speech** vector having a number of samples. A target excitation vector is calculated (102) in response to the present **speech** vector. An **error** value for each of a number of candidate excitation vectors stored in an overlapping table with the target excitation vector is calculated (106,104) by **repeating** a first portion of each of a group

of the candidate **speech** vectors at a second portion of each of the group so compensating for **speech** transitions such as between unvoiced and voiced regions...

...Information defining the location of the candidate excitation vector selected as having the smallest **error** value in the table and the filter coefficients for reproduction of the **speech** for the present **speech** vector are communicated (109...

...USE/ADVANTAGE - In **speech** synthesis. Adaptable to **speech** transitions esp. noticeable for women since fundamental frequencies that can be generated by women are...

...Abstract (Equivalent): A method of **encoding** **speech** based on determining sets of filter coefficients and corresponding excitation **frames** , said **speech** comprising **frames** each having a plurality of samples, comprising the steps of: determining (101) a set of filter coefficients of a filter in response to a present one of said **frames** of **speech** ; forming (102) a first excitation **frame** in response to the said present one of said **frames** of **speech** ; calculating (104, 106) an **error** value for each one of a plurality of candidate excitation **frames** stored in an adaptive code book in response to the said first excitation **frame** including forming virtual candidate excitation **frames** by **repeating** a first portion of each of a group of said candidate excitation **frames** at a second portion of said each of said group of said candidate excitation **frames** ; communicating (109) said filter coefficients and information defining the location of the candidate excitation **frame** selected as having the smallest **error** value in said adaptive code book to a decoder for reproduction of **speech** for the present **speech** **frame** , the said location defining information enabling the decoder to identify and itself form a virtual candidate excitation **frame** when said selected candidate excitation **frame** is a virtual excitation **frame** for the said present **speech** **frame** . (Dwg.1/9)c

...Abstract (Equivalent): The vocoder **encodes** **speech** using a code excited linear predicitive (CELP) **encoder** using a virtual searching technique during **speech** transistors e.g. from unvoiced to voiced regions of **speech** . The **encoder** compares candidate excitation vectors stored in a code book with a target excitation vector representing a **frame** of **speech** to determine the candidate vector that best matches the target vector by **repeating** a first portion of each candidate vector into a second portion of each candidate vector...

...For increased performance, a stochastically excited linear predictive (SELP) **encoder** is used in series with the adaptive CELP **encoder** . The SELP **encoder** is responsive to the difference between the target vector and the best matched candidate vector...

...that provides the best match. Both of the best matched candidate vectors are used in **speech** synthesis...

...USE - For **encoding** **speech** for communication to decoder for reproduction.

...Title Terms: **SPEECH** ;

International Patent Class (Main): **G10L-003/00** ...

... **G10L-009/14**

International Patent Class (Additional): **G10L-007/02** ...

19/3,K/9 (Item 9 from file: 350)
DIALOG(R)File 350:Derwent WPIX

004604565
WPI Acc No: 1986-107909/198617
XRPX Acc No: N86-079478

Cryptographic digital signal transceiver - uses hybrid sub-band coding and decoding with time delay compensation to reduce DSP on chip memory

Patent Assignee: ERICSSON GE MOBILE COMMUNICATIONS (TELF); GENERAL ELECTRIC CO (GENE); ERICSSON GE MOBILE COMMUNICATIONS INC (TELF)
Inventor: KAPPAGANTULA S; PETERSON E H; SZCZUTKOWSKI C F; ZINSER R L; KAPPAGANTU S; SZCZUTKOWS C F
Number of Countries: 009 Number of Patents: 013
Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
EP 178608	A	19860423				198617	B
JP 61098035	A	19860516	JP 85230046	A	19851017	198626	
US 4622680	A	19861111	US 84661598	A	19841017	198648	
US 4757536	A	19880712	US 84661733	A	19841017	198830	
CA 1249060	A	19890117				198910	
US 4817146	A	19890328	US 84661597	A	19841017	198915	
CA 1256178	A	19890620				198931	
CA 1258884	A	19890829				198939	
CA 1275700	C	19901030				199049	N
US 5051991	A	19910924	US 84661740	A	19841017	199141	
EP 178608	B1	19931229	EP 85112940	A	19851011	199401	
DE 3587710	G	19940210	DE 3587710	A	19851011	199407	
			EP 85112940	A	19851011		
KR 9404461	B1	19940525	KR 857610	A	19851016	199610	

Priority Applications (No Type Date): US 84661740 A 19841017; US 84661597 A 19841017; US 84661598 A 19841017; US 84661733 A 19841017
Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
EP 178608	A	E	85		
Designated States (Regional): DE FR GB NL SE					
EP 178608	B1	E	42	H04B-001/66	
Designated States (Regional): DE FR GB NL SE					
DE 3587710	G			H04B-001/66	Based on patent EP 178608
KR 9404461	B1			H04K-001/00	

...Abstract (Basic): of signal components. It is separated into frequency subbands of signal components which are separately **encoded** into binary-value digital signals that are then combined for transmission over a common digital communication channel. A hybrid **encoding** circuit **encodes** the signal components of the subbands in accordance with a predetermined **encoding** algorithm, especially an APCM circuit for adaptive pulse code modulation. It **encodes** the signal components of other subbands in accordance with a second predetermined **encoding** algorithm, especially a BCPCM circuit for block companded pulse code modulation...

...A circuit operates in the subband to time delay the digital signals in the **encoded** bit compressed digital format to provide a time delay while reducing digital memory requirement for effecting such delay. A control formats the transmitted digital data to include cryptographic synchronisation and **frame** synchronisation signals recurrently during an on going transmission so that late entry receipt of the...

...Abstract (Equivalent): subband outputs covering an overall frequency

band of approximately 180 to 2900 Hz; first digital **encoder** means (104,106,108) for receiving the three lowest frequency octave digital subband signals from said outputs and for **encoding** each such received subband in accordance with an APCM or ADPCM **encoding** algorithm to produce first **encoded** digital signal; second digital **encoder** means (110) for receiving the highest octave digital subband signals from another one of said outputs and for **encoding** such received subband signals in accordance with a BCPCM **encoding** algorithm to produce second **encoded** digital signals; and multiplex means (112) for combining said first and second **encoded** digital signals into a common string of digital signals for transmission over a digital signal...

...Abstract (Equivalent): delay compensation in the various subband channels is effected subsequent to a digital bandwidth compression **encoding** step on the transmitter side on the receiver side, similar time compensation may be provided...

...Hybrid subband coding and decoding (using different **encoding** /decoding algorithms in at least one subband channel) and subband time delay compensation at a...

...same time a special digital signal format is used so as to provide enhanced data **frame** synchronisation, enhanced cryptographic synchronisation and selective signalling ability within a cryptographic digital signal transceiver...

...synchronisation acquistion and ongoing synchronisation maintenance where the received digital signals are scanned for data **frames** succeeding the preamble portion, the data **frames** having a occuring in the initial preamble portion and also including a encrypted digital data. The **frame** synchronisation and the cryptographic synchronisation signals are **repeatedly** extracted so as to permit maintenance of **frame** synchronisation through-out the decoding of an encrypted message ...

...In the event **frame** synchronisation and/or cryptographic synchronisation are lost or not acquired from the preamble. The data **frames** are scanned and from which synchronisation, addressing and cryptographic synchronisation signals are nevertheless extracted. The ...

...ADVANTAGE - Permits full late entry and/or sync **recovery** even if synchronisation is never acquired from preamble or in event synchronisation is temporarily lost...

...A digitsed 180-2900 Hz **voice** band signal is divided into four octave-spaced subbands. The highest frequency subband (1450-2900 Hz) is **encoded** /decoded using a block companded pulse code modulation (BCPCM) algorithm while each of the lower frequency subbands is **encoded** /decoded using an adaptive pulse code modulation (APCM) or an adaptive differential pulse code modulation...

...The two **encoded** signals are combined in a multiplexer into a common string of digital signals for transmission...

...ADVANTAGE - Tends to maximise quality of transmitted **voice** signals while yet permitting implementation within constrained digital memory capacity and/or constrained transmission channel...

International Patent Class (Additional): **G10L-003/02** ...

19/3,K/10 (Item 10 from file: 347)

07535964 **Image available**
VOICE DECODING METHOD

PUB. NO.: 2003-029799 [JP 2003029799 A]
PUBLISHED: January 31, 2003 (20030131)
INVENTOR(s): YAMAURA TADASHI
TAZAKI HIROHISA
APPLICANT(s): MITSUBISHI ELECTRIC CORP
APPL. NO.: 2002-123468 [JP 2002123468]
Division of 09-060409 [JP 9760409]
FILED: March 14, 1997 (19970314)
PRIORITY: 08-276205 [JP 96276205], JP (Japan), October 18, 1996 (19961018)

VOICE DECODING METHOD

INTL CLASS: **G10L-019/12** ; H03M-007/36; H04L-001/00

ABSTRACT
PROBLEM TO BE SOLVED: To solve such problems that the quality of reproduced **audio** signals is low in a conventional waveform restoration method which **repeatedly** uses the parameters of a past **frame** when a coding **error** is generated and the delay becomes longer in a conventional method that conducts **interpolation** using the information of a future **frame** .

SOLUTION: In the **voice** decoding method, spectrum information of coded **voice** and **sound** source information are decoded in a **frame** unit and the **voice** are reproduced. When detection is made indicating that the decoding of the code is not correctly performed, the condition of the reproduced **voice** in a previous **frame** is judged and the reproduction restoration of the **voice** of a current **frame** is conducted in accordance with the judgment made above.

19/3,K/11 (Item 11 from file: 347)

05808094 **Image available**
METHOD OF **VOICE** DECODING AND DEVICE THEREFOR

PUB. NO.: 10-091194 [JP 10091194 A]
PUBLISHED: April 10, 1998 (19980410)
INVENTOR(s): IIJIMA KAZUYUKI
NISHIGUCHI MASAYUKI
MATSUMOTO ATSUSHI
APPLICANT(s): SONY CORP [000218] (A Japanese Company or Corporation), JP (Japan)
APPL. NO.: 08-246679 [JP 96246679]
FILED: September 18, 1996 (19960918)

METHOD OF **VOICE** DECODING AND DEVICE THEREFOR

INTL CLASS: **G10L-009/14** ; **G10L-009/18** ; H03M-007/30
...JAPIO KEYWORD: **Speech** Recognition & Synthesis)

ABSTRACT

...prevent an unnatural feeling from being generated, in which a long cycle pitch of a **frame** is generated in a voiceless **sound** **frame** in which no pitch should exist, if a same parameter is **repeatedly** used in a voiceless **sound** **error** **frame** .

...

...SOLUTION: In a case of decoding an **encoded** **speech** signal obtained by waveform- **encoding** a time axis waveform signal of each **encoding** unit obtained by dividing an input **speech** signal into prescribed **encoding** units on a time axis, CRC codes of input data are checked by CRC(cyclic redundancy check) check and a bad **frame** masking circuit 281, and a **frame** with **error** is processed with a bad **frame** masking so that the **frame** parameter of the immediately preceding **frame** is used again, and when the erroneous **frame** is a voiceless **sound** **frame** , a voiceless **sound** synthesis part 2 adds noise to a driving vector from a noise coding note or

19/3,K/12 (Item 12 from file: 347)

DIALOG(R)File 347:JAPIO

05604849 **Image available**

VARIABLE RATE **ENCODING** SYSTEM

PUB. NO.: 09-219649 [JP 9219649 A]
PUBLISHED: August 19, 1997 (19970819)
INVENTOR(s): KAWAHARA NOBUAKI
SASAKI SEIJI
URABE KENZO
APPLICANT(s): KOKUSAI ELECTRIC CO LTD [000112] (A Japanese Company or Corporation), JP (Japan)
APPL. NO.: 08-048402 [JP 9648402]
FILED: February 13, 1996 (19960213)

VARIABLE RATE **ENCODING** SYSTEM

INTL CLASS: H03M-007/30; **G10L-009/18**
...JAPIO KEYWORD: **Speech** Recognition & Synthesis)

ABSTRACT

PROBLEM TO BE SOLVED: To provide a variable rate **encoding** system which can be applied to the communication system by an adaptive modulation system changing...

... transmission rate according to the change of the state of a propagation path and makes **speech** quality difficult to be affected by the influence caused by the change of the transmission...

...SOLUTION: After a **sound** **encoding** is performed for an input signal by a fixed **encoding** rate on a transmission side, the **redundant** bit of an **error** correction is added to the input signal by the **encoding** ratio corresponding to a transmission rate in a variable rate communication path **encoder** 22 and the input signal is inputted in an adaptive transmission formatter 23. The adaptive transmission formatter 23 stores the transmission information on past few **frames** , holds the **redundant** bit of the present **frame** and transmits only information bit when the state of a propagation path is poor, and adds the **redundant** bit of the **frame** just

before in which the state of the propagation path is poor to the **redundant** bit of the present **frame** and transmits the **redundant** bit when the state of propagation path is excellent. On a reception side, the **redundant** bit is returned to the original **frame** and the bit is decoded.

19/3,K/13 (Item 13 from file: 347)
DIALOG(R)File 347:JAPIO

03202133 **Image available**
SOUND ENCODER

PUB. NO.: 02-177633 [JP 2177633 A]
PUBLISHED: July 10, 1990 (19900710)
INVENTOR(s): TAGUCHI SATORU
APPLICANT(s): NEC CORP [000423] (A Japanese Company or Corporation), JP (Japan)
APPL. NO.: 01-226791 [JP 89226791]
FILED: September 01, 1989 (19890901)
JOURNAL: Section: E, Section No. 983, Vol. 14, No. 448, Pg. 66, September 26, 1990 (19900926)

SOUND ENCODER

INTL CLASS: H04L-001/00; **G10L-009/18** ; H04B-014/04; H04L-001/08
...JAPIO KEYWORD: **Speech** Recognition & Synthesis)

ABSTRACT

PURPOSE: To obtain a **sound encoder** , which is immune to the **error** of a transmission line by providing a means to decide a type in a synthesization ...

...CONSTITUTION: When a **frame** is composed of A, the whole or one part from the second bit to the 160th bit in the **frame** is uncorrelated to the part from the 161st bit to the 320th bit. On the other hand, when the **frame** is composed of B, the whole or one part from the second bit to the 160th bit in the **frame** is completely correlated to the part from the 161st bit to the 320th bit (in case that there is no **error** of the transmission line) or is strongly correlated (in case that there is the **error** of the transmission line). Thus, **frame** constitution can be easily detected according to the presence and absence of this correlation. Thus, the rate of the **redundant** bit can be made variable suitably for the various condition of the different degree of a code **error** .
?

22/3,K/1 (Item 1 from file: 350)
DIALOG(R)File 350:Derwent WPIX

003007277
WPI Acc No: 1981-A7283D/198104

TDMA satellite communications system - uses spare TDMA time slots in each frame **sequence to increase rain margin during fading**

Patent Assignee: WESTERN ELECTRIC CO INC (AMTT)
Inventor: ACAMPORA A
Number of Countries: 005 Number of Patents: 006
Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
WO 8100034	A	19810108				198104	B
EP 30551	A	19810624	EP 80901307	A	19800613	198127	
JP 56500733	W	19810528				198150	
US 4309764	A	19820105				198204	
EP 30551	B	19840926				198439	
DE 3069273	G	19841031				198445	

Priority Applications (No Type Date): US 7951022 A 19790622
Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
WO 8100034	A	E			
Designated States (National): JP					
Designated States (Regional): DE FR GB					
EP 30551	A	E			
Designated States (Regional): DE FR GB					
EP 30551	B	E			
Designated States (Regional): DE FR GB					

... **uses spare TDMA time slots in each** frame **sequence to increase rain margin during fading**

...Abstract (Basic): The **transmitter** is capable of launching one or more message bursts during a **frame sequence** to a **number** of receivers. Each message burst includes preamble and data which are received from temestrial lines by a circuit (20) which formets the received signals into the proper digital arrangement. A **first** memory (32) stores a further portion of the preamble. A **multiplexer** (36) **multiplexes** the information received and that stored in the **first** memory to form a message burst for **transmission** to a receiver...

...A switch controller (27) generates a **first** or **second** control signal according to the existance of non-fade or fade conditions between the **transmitter** and the receiver. An **encoder** (24) **encodes** the formatted input signal into a redundancy- **encoded** output signal. A **second** memory (30) stores an extended version of the remaining portion of the preamble information stored in the **first** memory (32). Switches (25,38) couple both the input to the **transmitter** and the **first** memory (32) to the **multiplexer** (36) in response to the **first** control signal from the controller (27). The switches couple both the **encoder** (24) and the **second** memory (30) to the **multiplexer** (36) in response to the **second0** control signal from the switch controller forming a message burst at the output of the **multiplexer** .

...Abstract (Equivalent): The satellite communications system includes a **transmitter** receiving an input signal made up of data and part of the preamble information associated with each message burst to be sent to a receiver. A **first** memory stores the non-fade preamble and postamble.

A **multiplexer** **multiplexes** that stored information with the input signal to the **transmitter** , to form a message burst for **transmission** .

...

...A switch controller generates a **first** control signal in response to a non-fade condition prevailing in **transmission** between **transmitter** and receiver of the message burst to be sent. It generates a **second** control signal in response to a detected fade condition. An **encoder** **encodes** the input signal to form a redundantly **encoded** output signal. A **second** memory stores an extended fade preamble and postamble. Switches couple the appropriate memory and **encoder** to the **multiplexer** in accordance with the **first** or **second** control signal from the switch controller

...Title Terms: **FRAME** ;

?

24/3,K/1 (Item 1 from file: 350)
DIALOG(R)File 350:Derwent WPIX

013387671 **Image available**
WPI Acc No: 2000-559609/200052
XRPX Acc No: N00-414176

Error **detection** encoder **for speech** encoding **using cyclic redundancy check without using an increased number of** redundant error **check bits**

Patent Assignee: NEC CORP (NIDE)
Inventor: ITO H; SERIZAWA M
Number of Countries: 012 Number of Patents: 005
Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
EP 1028555	A2	20000816	EP 2000250039	A	20000209	200052	B
CA 2298221	A1	20000810	CA 2298221	A	20000209	200052	
JP 2000232374	A	20000822	JP 9932122	A	19990210	200055	
JP 3257534	B2	20020218	JP 9932122	A	19990210	200215	
US 6532564	B1	20030311	US 2000499218	A	20000207	200321	

Priority Applications (No Type Date): JP 9932122 A 19990210
Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
EP 1028555	A2	E	12	H04L-001/00	
Designated States (Regional): AL DE FR GB LT LV MK RO SI					
CA 2298221	A1	E		H04L-001/22	
JP 2000232374	A		11	H03M-013/00	
JP 3257534	B2		10	H03M-013/09	Previous Publ. patent JP 2000232374
US 6532564	B1			H03M-013/00	

Error **detection** encoder **for speech** encoding **using cyclic redundancy check without using an increased number of** redundant error **check bits**

Abstract (Basic):

... The **encoder** includes separation circuitry for separating an input signal into a two sequences of **error** protected bits. Calculation circuitry produces an **error** check sequence from the first sequence and concatenates the **error** check sequence to the first sequence to produce a third sequence. The second sequence may...

... A **multiplexer** is provided for segmenting the third **sequence** into a **number** of first blocks and segmenting the first sub-sequence into several second blocks corresponding to the first blocks and **multiplexing** each of the first blocks with a corresponding one of the second blocks to produce...

...second sub-sequence is concatenated to the fourth sequence to produce an output sequence for **transmission** .

...

...For speech **encoding** using cyclic redundancy check (CRC...

...Improved detection of **errors** in speech **encoding** without using an increased number of **redundant** **error** check bits...

...The drawing shows a schematic diagram of the **error** detection **encoder** .

Title Terms: **ERROR** ;

24/3,K/2 (Item 2 from file: 350)
DIALOG(R)File 350:Derwent WPIX

012050149 **Image available**
WPI Acc No: 1998-467059/199840
XRPX Acc No: N98-363890

Apparatus for recording and reproducing digital audio and video signals on slanted track magnetic tape - uses periodicity based sample number allocation and error **rate detection with selection of sequence signal for slanted track reproduction**

Patent Assignee: SONY CORP (SONY)
Inventor: SATO Y
Number of Countries: 001 Number of Patents: 001
Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
US 5796912	A	19980818	US 9374293	A	19930609	199840	B

Priority Applications (No Type Date): JP 92178829 A 19920615
Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
US 5796912	A		19	G11B-005/02	

... **uses periodicity based sample number allocation and** error **rate detection with selection of sequence signal for slanted track reproduction**

...Abstract (Basic): to an A/D converter [14]. Linear quantised digital data in both cases is then **transmitted** to a **encoders** [17,9] and then supplied to a **multiplexer** [18...

...The **multiplexed** data is amplified by a recording amplifier [10] and passed to a rotary recording head...

...Reproduction from the tape is performed via the playback head for video and audio decoding. **Error** status signals are sent to the **error** counter [4] and then to the selector [5] which produces a selection control signal. This...

...information is used during decoding of the audio data as signals for discriminating the field **sequence** of the **numbers** of samples of the audio data...

...ADVANTAGE- Recording information is obtained stably so that highly reliable editing can be performed. **Redundant** playback circuitry may be eliminated if only recording is desired to enable the structure to ...

...Title Terms: **ERROR ;**
?

27/3,K/1 (Item 1 from file: 350)
DIALOG(R)File 350:Derwent WPIX

010443487 **Image available**
WPI Acc No: 1995-344806/199544
XRPX Acc No: N95-257686

Modem using pilot signals for equalisation and frame **synchronisation - converts received signals to complex baseband signal and uses pilot signal to mark start of each symbol to maintain synchronisation and provide good** signal **to** noise ratio

Patent Assignee: GLENAYRE ELECTRONICS INC (GLEN-N)
Inventor: FAWCETT G S; MARCHETTO R F; STEWART T A
Number of Countries: 062 Number of Patents: 007
Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
WO 9526081	A1	19950928	WO 95US3415	A	19950317	199544	B
AU 9521233	A	19951009	AU 9521233	A	19950317	199603	
EP 761043	A1	19970312	EP 95914107	A	19950317	199715	
			WO 95US3415	A	19950317		
US 5666378	A	19970909	US 94215129	A	19940318	199742	
KR 97701951	A	19970412	WO 95US3415	A	19950317	199817	
			KR 96705165	A	19960918		
US 5787133	A	19980728	US 94215129	A	19940318	199837	
			US 95474332	A	19950607		
CN 1153580	A	19970702	CN 95192686	A	19950317	200306	
			WO 95US3415	A	19950317		

Priority Applications (No Type Date): US 94215129 A 19940318; US 95474332 A 19950607

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
WO 9526081	A1	E	44	H04B-001/38	

Designated States (National): AM AT AU BB BG BR BY CA CH CN CZ DE DK EE ES FI GB GE HU JP KE KG KP KR KZ LK LR LT LU LV MD MG MN MW MX NL NO NZ PL PT RO RU SD SE SG SI SK TJ TT UA UZ VN
Designated States (Regional): AT BE CH DE DK ES FR GB GR IE IT KE LU MC MW NL OA PT SD SE SZ UG

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
AU 9521233	A			H04B-001/38	Based on patent WO 9526081
EP 761043	A1	E	44	H04B-001/38	Based on patent WO 9526081

Designated States (Regional): CH DE DK ES FR GB SE

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
US 5666378	A		20	H04B-001/38	
KR 97701951	A			H04B-001/38	Based on patent WO 9526081
US 5787133	A			H04L-007/04	Div ex application US 94215129; Div ex patent US 5666378
CN 1153580	A			H04B-001/38	Based on patent WO 9526081

Modem using pilot signals for equalisation and frame **synchronisation...**

...**uses pilot signal to mark start of each symbol to maintain synchronisation and provide good** signal **to** noise ratio

...Abstract (Basic): The signal demodulator for a signal including data symbols used to **encode** data and pilot symbols. the demodulator has a sampling circuit which samples a received signal...

...synchronised signal, producing a data symbol signal. A decoder decodes the data symbol signal and **recovers** the data corresp to the data symbols...

...Abstract (Equivalent): Apparatus for demodulating an input signal that includes data symbols used to **encode** data and pilot symbols, said

pilot symbols being a predetermined pseudo-random sequence of digital signals that are interspersed one at a time with said data symbols, so that each **repeated** sequence of said predetermined pseudo-random sequence of pilot symbols defines one data **frame** of a data message that is collectively established by said data symbols, said demodulator comprising...

...synchronization means including means for processing the complex baseband signal to determine a symbol timing **error** signal, said synchronization means further including means responsive to said symbol timing **error** signal for adjusting symbol sampling time to produce a synchronized signal in which signal samples...

...e) **frame** synchronization means for receiving the signal samples that correspond to said input pilot symbols and...

...of digital signals that represent said predetermined pseudo-random sequence of said pilot symbols, said **frame** synchronization means detecting the end of one data **frame** and the beginning of a next data **frame** by determining processing by said **frame** synchronization means of the signal sample that corresponds to the last pilot symbol of said ...

...to the first pilot symbol of said predetermined pseudo-random sequence of pilot symbols, said **frame** synchronization means supplying a signal to indicate detection of the end of one data **frame** and the start of a next data **frame** , said **frame** synchronization means processing said signal samples that correspond to said pilot symbols to determine an autocorrelation value and supplying said signal to indicate detection of the end of one data **frame** and the start of a next data **frame** when said autocorrelation value exceeds a predetermined threshold value ...

...Title Terms: **FRAME** ;

?

29/3,K/1 (Item 1 from file: 350)
DIALOG(R)File 350:Derwent WPIX

009676917 **Image available**
WPI Acc No: 1993-370470/199347
Related WPI Acc No: 1993-160893; 1994-169750; 1994-310830; 1997-022815
XRPX Acc No: N93-286034

Magnetic resonance cine flow imaging appts. e.g. for cardiac, angiography, and circulatory examination - dividing positive and negative portions of K-space into n segments, generating groups of echo sequences in each cardiac cycle, and dividing into groups of n contiguous echoes.

Patent Assignee: PICKER INT INC (PXRM)
Inventor: NESSAIVER M S; MURDOCH J B
Number of Countries: 004 Number of Patents: 005
Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
EP 571071	A1	19931124	EP 93302041	A	19930317	199347	B
US 5329925	A	19940719	US 91791855	A	19911114	199428	
			US 92859153	A	19920327		
			US 92874807	A	19920428		
US 5447155	A	19950905	US 91791855	A	19911114	199541	N
			US 92859153	A	19920327		
EP 571071	B1	19990915	EP 93302041	A	19930317	199942	
DE 69326379	E	19991021	DE 626379	A	19930317	199950	
			EP 93302041	A	19930317		

Priority Applications (No Type Date): US 92874807 A 19920428; US 91791855 A 19911114; US 92859153 A 19920327

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
EP 571071	A1	E	19	G01R-033/56	
Designated States (Regional): DE FR NL					
US 5329925	A		18	A61B-005/055	CIP of application US 91791855
					CIP of application US 92859153
					CIP of patent US 5273040
US 5447155	A		11	A61B-005/055	CIP of application US 91791855
					CIP of patent US 5273040
EP 571071	B1	E		G01R-033/56	
Designated States (Regional): DE FR NL					
DE 69326379	E			G01R-033/56	Based on patent EP 571071

...Abstract (Basic): positive and negative portions of K-space which is segmented into corresponding n segments. The **first** segment of both positive and negative K-space contains views with the highest order frequency components, **subsequent** segments containing views with progressively lower order frequency components. The...

...group corresponds to a different time interval of the subjects cardiac cycle, and reconstructing a **frame** image representation from the echo signals of each group...

```
File   2:INSPEC 1969-2004/Mar W1
         (c) 2004 Institution of Electrical Engineers
File   6:NTIS 1964-2004/Mar W2
         (c) 2004 NTIS, Intl Cpyrght All Rights Res
File   8:Ei Compendex(R) 1970-2004/Mar W1
         (c) 2004 Elsevier Eng.  Info. Inc.
File  34:SciSearch(R) Cited Ref Sci 1990-2004/Mar W1
         (c) 2004 Inst for Sci Info
File  35:Dissertation Abs Online 1861-2004/Feb
         (c) 2004 ProQuest Info&Learning
File  65:Inside Conferences 1993-2004/Mar W2
         (c) 2004 BLDSC all rts. reserv.
File  94:JICST-EPlus 1985-2004/Mar W1
         (c)2004 Japan Science and Tech Corp(JST)
File  95:TEME-Technology & Management 1989-2004/Feb W5
         (c) 2004 FIZ TECHNIK
File  99:Wilson Appl. Sci & Tech Abs 1983-2004/Feb
         (c) 2004 The HW Wilson Co.
File 144:Pascal 1973-2004/Mar W1
         (c) 2004 INIST/CNRS
File 233:Internet & Personal Comp. Abs. 1981-2003/Sep
         (c) 2003 EBSCO Pub.
File 434:SciSearch(R) Cited Ref Sci 1974-1989/Dec
         (c) 1998 Inst for Sci Info
File 583:Gale Group Globalbase(TM) 1986-2002/Dec 13
         (c) 2002 The Gale Group
File 603:Newspaper Abstracts 1984-1988
         (c)2001 ProQuest Info&Learning
File 483:Newspaper Abs Daily 1986-2004/Mar 12
         (c) 2004 ProQuest Info&Learning

Set     Items   Description
S1     931621   VOICE OR AUDIO OR SOUND OR SPEECH
S2     953550   FRAME?
S3      30973   (INTERPOLAT? OR ENCOD?) AND (REPEAT? OR ITERATIV? OR REDUN-
             DANT? OR REITERA?)
S4       4098   S3 AND (RECOVER? OR ERROR?)
S5     109226   SIGNAL(3N)NOISE()RATIO
S6       5750   VOIP OR VOICE(3N)INTERNET
S7       1842   S2 AND (FIRST OR INITIAL?) AND (SECOND OR SUBSEQUENT?) AND
             ENCOD?
S8      11825   PARAMETER? AND PACKET?
S9      13248   CONSONANT??
S10     35514   (INDEX OR SEQUENCE)(3N)NUMBER??
S11        79   S10 AND (MULTIPLEX? OR MULTI()PLEX?) AND (TRANSMIT? OR TRA-
             NSMIS? OR SEND OR SENDING OR SENDS)
S12    153783   (SAME OR CLOSE OR EQUAL OR EQUIVALENT OR APPROXIMAT? OR MA-
             XIMUM OR HIGHEST) AND MATCH?
S13       326   AU=(AMANO, F? OR AMANO F?)
S14       584   (S1 OR S6) AND S3
S15       584   S14 AND S3
S16         0   S15 AND S8 AND S9
S17        12   S15 AND (S8 OR S9)
S18        12   RD S17 (unique items)
S19       122   S9 AND S12 AND (S1 OR S6)
S20         2   S19 AND S5
S21         2   RD S20 (unique items)
S22       437   S2 AND S4
S23        39   S22 AND S1
S24         1   S23 AND S8
S25         0   S24 NOT (S17 OR S20)
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S26	0	S23 AND S9
S27	0	S23 AND S10
S28	5	S23 AND S5
S29	5	S28 NOT (S17 OR S20)
S30	5	RD S29 (unique items)
S31	41	S7 AND S12
S32	0	S31 AND S11
S33	0	S31 AND S10
S34	3	S31 AND S1
S35	3	S34 NOT (S28 OR S17 OR S20)
S36	3	RD S35 (unique items)
S37	70	S13 AND (S1 OR S6)
S38	3	S37 AND S2
S39	3	S38 NOT (S34 OR S28 OR S17 OR S20)
S40	2	RD S39 (unique items)
S41	0	S37 AND S9
S42	0	S37 AND S8

18/3,K/1 (Item 1 from file: 2)
DIALOG(R)File 2:INSPEC

7603591 INSPEC Abstract Number: B2003-06-6220M-010
Title: Bandwidth efficient AMR operation for VoIP
Author(s): Johansson, I.; Frankkila, T.; Synnergren, P.
Author Affiliation: Multimedia Technol., Ericsson AB, Lulea, Sweden
Conference Title: 2002 IEEE Speech Coding Workshop. Proceedings (Cat. No.02EX592) p.150-2
Publisher: IEEE, Piscataway, NJ, USA
Publication Date: 2002 Country of Publication: USA ix+184 pp.
ISBN: 0 7803 7549 1 Material Identity Number: XX-2002-03143
U.S. Copyright Clearance Center Code: 0-7803-7549-1/02/$17.00
Conference Title: 2002 IEEE Speech Coding Workshop. Proceedings
Conference Date: 6-9 Oct. 2002 Conference Location: Ibaraki, Japan
Language: English
Subfile: B

Title: Bandwidth efficient AMR operation for VoIP
Abstract: An example of a bandwidth efficient adaptive multi rate (AMR) system for **Voice** over IP (**VoIP**) is presented. In **VoIP , packet** losses cause degradation of the synthesized **speech** . The distortions may propagate over several consecutive frames, since predictors in the codec exploit inter-frame correlations to gain coding efficiency. To reduce the effects of **packet** loss, forward error correction (FEC) that adds **redundant** information to **voice packets** can be used. However, while FEC can reduce the effects of **packet** loss, it will increase the amount of bandwidth used by the **voice** stream, which is not desirable. In this paper we propose FEC methods like partial redundancy, selective redundancy for the most sensitive frames and **parameter interpolation** in conjunction with AMR codec mode adaptation, which secure the **speech** quality when using AMR for **VoIP** without increasing the bandwidth substantially.
...Descriptors: **interpolation** ; ...

... **speech** codecs
...Identifiers: **VoIP** ; ...

... **packet** losses...

...synthesized **speech** ; ...

... **parameter interpolation** ; ...

... **speech** quality

18/3,K/2 (Item 2 from file: 2)
DIALOG(R)File 2:INSPEC

7241038 INSPEC Abstract Number: B2002-05-6130C-011
Title: An interpolative **decoding approach for** speech **streaming services and** voice **over IP**
Author(s): Fingscheidt, T.; Prez, J.F.G.
Author Affiliation: Mobile Phones (ICM MP), Siemens AG, Munich, Germany
Journal: ITG-Fachbericht Conference Title: ITG-Fachber. (Germany) no.170 p.353-6
Publisher: VDE-Verlag,

Publication Date: 2002 Country of Publication: Germany
CODEN: ITGFEY ISSN: 0932-6022
SICI: 0932-6022(2002)170L.353:IDAS;1-Z
Material Identity Number: G434-2002-001
Conference Title: 4th International ITG Conference Source and Channel Coding
Conference Date: 28-30 Jan. 2002 Conference Location: Berlin, Germany
Language: English
Subfile: B

Title: An interpolative **decoding approach for** speech **streaming services and** voice **over IP**

Abstract: **Voice** -over-IP services usually require a receiver buffer to overcome the effect of delay jitter and lost **packets** . In streaming services ARQ schemes (automatic **repeat** request) are used to retrieve the lost **packets** . Since streaming as a non-conversational service has relaxed delay constraints, a comparably large buffer...

... an important means for achieving robustness in these two application scenarios. When frames of coded **speech** are buffered at the receiver (pre-decoder buffering), the **speech** coder **parameters** of lost frames can be computed by **interpolation** rather than extrapolation. Modern **speech** coders rely heavily on predictive quantization which makes **interpolation** a difficult task. We present a solution to **interpolation** of predictively coded **parameters** . Significant gains are achievable over GSM channels as well as random frame loss channels.

...Descriptors: **interpolation** ; ...

... **speech** coding

Identifiers: **interpolative** decoding...

... **speech** streaming services...

... **voice** over IP...

...coded **speech** ; ...

... **speech** coder...

... **interpolation** ; ...

...predictively coded **parameters** ;

18/3,K/3 (Item 3 from file: 2)

DIALOG(R)File 2:INSPEC

03624034 INSPEC Abstract Number: B90030371, C90031971

Title: Direct coding of a class of non-stationary signals based on mixed transforms

Author(s): Mikhael, W.B.; Spanias, A.S.
Author Affiliation: Dept. of Electr. & Comput. Eng., West Virginia Univ., Morgantown, WV, USA

Conference Title: 1989 IEEE International Symposium on Circuits and Systems (Cat. No.89CH2692-2) p.280-3 vol.1
Publisher: IEEE, New York, NY, USA
Publication Date: 1989 Country of Publication: USA 3 vol. xl+2246 pp.
U.S. Copyright Clearance Center Code: CH2692-2/89/0000-0280$01.00
Conference Sponsor: IEEE

Conference Date: 8-11 May 1989 Conference Location: Portland, OR, USA
Language: English
Subfile: B C

...Abstract: are used, namely, the discrete Fourier transform (DFT) and the Walsh-Hadamard transform (WHT). An **iterative** algorithm is used to implement the mixed representation. A finite number of DFT components is used to represent the selective (narrowband) dominant spectrum of **speech** on a frame-by-frame basis. The whitened (broadband) residual signal is represented using a small number of Walsh components. The **parameters** of the mixed representation are coded directly by coding the DFT and WHT components. Results are given for frame-by-frame and **packet** coding at rates ranging from 16 kb/s to 9.6 kb/s.
Descriptors: **encoding** ; ...

... **iterative** methods
...Identifiers: **speech** spectrum...

... **iterative** algorithm...

... **packet** coding

18/3,K/4 (Item 4 from file: 2)
DIALOG(R)File 2:INSPEC

03569178 INSPEC Abstract Number: B90016888, C90011223
Title: Reduced bit-rate representation of speech **using mixed Fourier-Walsh transforms**
Author(s): Mikhael, W.B.; Spanias, A.S.
Author Affiliation: Dept. of Electr. Eng. & Commun. Sci., Univ. of Central Florida, Orlando, FL, USA
Conference Title: Conference Record. Twenty-Second Asilomar Conference on Signals, Systems and Computers (Cat. No.88CH2835-7) p.366-70 vol.1
Publisher: Maple Press, San Jose, CA, USA
Publication Date: 1989 Country of Publication: USA 2 vol. xviii+985 pp.
U.S. Copyright Clearance Center Code: 22ACSSC-12/88/0366$01.00
Conference Sponsor: Naval Postgraduate School; San Jose State Univ
Conference Date: 31 Oct.-2 Nov. 1988 Conference Location: Pacific Grove, CA, USA
Language: English
Subfile: B C

Title: Reduced bit-rate representation of speech **using mixed Fourier-Walsh transforms**
...Abstract: are used, namely the discrete Fourier transform (DFT) and the Walsh-Hadamard transform (WHT). An **iterative** algorithm is used to implement the mixed representation. A finite number of DFT components is used to represent the selective (narrowband) dominant spectrum of **speech** on a frame-by-frame basis. The whitened (broadband) residual signal is represented using a small number of Walsh components. The **parameters** of the mixed representation are coded directly by coding the DFT and WHT components. Results are given for frame-by-frame and **packet** coding at rates ranging from 16 to 9.6 kb/s.
...Descriptors: **encoding** ; ...

... **speech** analysis and processing
Identifiers: reduced bit-rate representation of **speech** ; ...

... **iterative** algorithm...

... **packet** coding

18/3,K/5 (Item 5 from file: 2)
DIALOG(R)File 2:INSPEC

03533733 INSPEC Abstract Number: C90005349
Title: On-line recognition of vocalised Pitman shorthand outlines
Author(s): Leedham, C.G.; Qiao, Y.
Author Affiliation: Dept. of Electron. Syst. Eng., Essex Univ., Colchester, UK
Conference Title: IEE Colloquium on 'Character Recognition and Applications' (Digest No.109) p.10/1-5
Publisher: IEE, London, UK
Publication Date: 1989 Country of Publication: UK 60 pp.
Conference Sponsor: IEE
Conference Date: 2 Oct. 1989 Conference Location: London, UK
Language: English
Subfile: C

Abstract: The obvious method of achieving simultaneous transcription of **speech** would be by using a **speech** recogniser. Unfortunately, the difficulties involved in the recognition of unconstrained, unlimited vocabulary connected **speech** are daunting and will not be solved for many years. Therefore, alternative methods have been...

... require a human intermediary to carry out the computationally difficult tasks of isolating the speakers **voice** from other background sounds, separating words in the **speech** , ignoring the **redundant** **speech** utterances and simultaneously **encoding** the essential **speech** components of each word via some other medium such as keyboard or handwriting. The authors describe an approach for the segmentation and classification of the **consonant** part of a vocalised Pitman outline into its phonetic features involves three stages. Each segment...
...Identifiers: **speech** transcription...

... **consonant** ;

18/3,K/6 (Item 1 from file: 34)
DIALOG(R)File 34:SciSearch(R) Cited Ref Sci

10650162 Genuine Article#: 551WF No. References: 16
Title: Imitation of nonwords by deaf children after cochlear implantation: Preliminary findings
Author(s): Cleary M (REPRINT) ; Dillon C; Pisoni DB
Corporate Source: Indiana Univ,Dept Psychol, Speech Res Lab,1101 E 10th St/Bloomington//IN/47405 (REPRINT); Indiana Univ,Dept Psychol, Speech Res Lab,Bloomington//IN/47405
Journal: ANNALS OF OTOLOGY RHINOLOGY AND LARYNGOLOGY, 2002, V111, N5,2,189 (MAY), P91-96
ISSN: 0003-4894 Publication date: 20020500
Publisher: ANNALS PUBL CO, 4507 LACLEDE AVE, ST LOUIS, MO 63108 USA
Language: English Document Type: ARTICLE (ABSTRACT AVAILABLE)

...Abstract: judgments of repetition accuracy. The results revealed wide variability in the children's ability to **repeat** the novel **sound** sequences. Individual differences in the component processes of **encoding** , memory, and **speech** production were strongly reflected in the nonword repetition scores. Duration of deafness before implantation also appeared to be a factor associated with imitation performance. Linguistic analyses of the initial **consonants** in the nonwords revealed that coronal stops were imitated best, followed by the coronal fricative...
...Identifiers--SHORT-TERM-MEMORY; VOCABULARY DEVELOPMENT; **SPEECH**

18/3,K/7 (Item 2 from file: 34)
DIALOG(R)File 34:SciSearch(R) Cited Ref Sci

07510205 Genuine Article#: 174PK No. References: 57
Title: Effects of talker, rate, and amplitude variation on recognition memory for spoken words
Author(s): Bradlow AR (REPRINT) ; Nygaard LC; Pisoni DB
Corporate Source: NORTHWESTERN UNIV,DEPT LINGUIST, 2016 SHERIDAN RD/EVANSTON//IL/60208 (REPRINT); EMORY UNIV,/ATLANTA//GA/30322; INDIANA UNIV,/BLOOMINGTON//IN/47405
Journal: PERCEPTION & PSYCHOPHYSICS, 1999, V61, N2 (FEB), P206-219
ISSN: 0031-5117 Publication date: 19990200
Publisher: PSYCHONOMIC SOC INC, 1710 FORTVIEW RD, AUSTIN, TX 78704
Language: English Document Type: ARTICLE (ABSTRACT AVAILABLE)

Abstract: This study investigated the **encoding** of the surface form of spoken words using a continuous recognition memory task. The purpose...

...or ''new.'' Listeners were more accurate at recognizing a word as old if it was **repeated** by the same talker and at the same speaking rate; however there was no recognition advantage for words **repeated** at the same overall amplitude. In Experiment 2, listeners were first asked to judge whether...
...old or new, as before, and then they had to explicitly judge whether it was **repeated** by the same talker, at the same rate, or at the same amplitude. On the first task, listeners again showed an advantage in recognition memory for words **repeated** by the same talker and at same speaking rate, but no advantage occurred for the...

...in all three conditions, listeners were able to explicitly detect whether an old word was **repeated** by the same talker, at the same rate, or at the same amplitude. These data suggest that although information about all three properties of spoken words is **encoded** and retained in memory, each source of stimulus variation differs in the extent to which...
...Identifiers--LONG-TERM-MEMORY; SPEAKING RATE; STIMULUS VARIABILITY; **SPEECH** -PERCEPTION; CONVERSATIONAL **SPEECH** ; STOP **CONSONANTS** ; SPEAKERS **VOICE** ; ARTICULATION; INFORMATION; HEARING

18/3,K/8 (Item 3 from file: 34)
DIALOG(R)File 34:SciSearch(R) Cited Ref Sci

05035646 Genuine Article#: TK761 No. References: 65
Title: SYLLABLE STRUCTURE IN SPEECH **PRODUCTION - ARE SYLLABLES CHUNKS OR SCHEMAS**

Author(s): SEVALD CA; DELL GS; COLE JS
Corporate Source: INST RES COGNIT SCI,3401 WALNUT ST,SUITE 400
C/PHILADELPHIA//PA/19104; UNIV ILLINOIS/URBANA//IL/61801
Journal: JOURNAL OF MEMORY AND LANGUAGE, 1995, V34, N6 (DEC), P807-820
ISSN: 0749-596X
Language: ENGLISH   Document Type: ARTICLE   (Abstract Available)

Title: SYLLABLE STRUCTURE IN SPEECH **PRODUCTION - ARE SYLLABLES CHUNKS OR SCHEMAS**

Abstract: Theories of **speech** production hold divergent views of the syllable. Some theories do not use syllables at all...

...chunks that specify their phonological content or schemas that specify an abstract structure (e.g. **consonant** -vowel- **consonant**). In three experiments, speakers were asked to **repeat** pairs of phonological words as often as possible in a 4-s period. **Speech** rate was faster when both the structure and content of the first phonological word were **repeated** in the first syllable of the second one, compared to a condition in which all or most of the sounds were **repeated** but the structure was not. There was no additional advantage for **repeating** both content and structure over **repeating** structure alone. The results support the view that syllable structure is separable from phonemic content...

Research Fronts: 94-3281 003 (AUDIOVISUAL **SPEECH** ; STRESS SYSTEMS IN LANGUAGE; PHONETIC CONTEXT; PROSODIC PHONOLOGY; LOCUS EQUATIONS; ARTICULATORY OVERLAP; PHONEMIC DISTINCTIONS)
94-3315 003 (LEXICAL ACCESS IN **SPEECH** PRODUCTION; SEMANTIC ERRORS; WRITTEN WORD COMPREHENSION; PHONOLOGICAL **ENCODING** ; DEEP DYSPHASIA; APHASIC NAMING)

18/3,K/9 (Item 4 from file: 34)
DIALOG(R)File 34:SciSearch(R) Cited Ref Sci

02946486 Genuine Article#: MU334 No. References: 36
Title: PRECISION AND ACCURACY OF SUBJECTIVE TIME-ESTIMATION IN DIFFERENT MEMORY DISORDERS
Author(s): NICHELLI P; VENNERI A; MOLINARI M; TAVANI F; GRAFMAN J
Corporate Source: NINCDS,MNB,COGNIT NEUROSCI SECT,BLDG 10,ROOM 5S209/BETHESDA//MD/20892; UNIV MODENA,NEUROL CLIN/I-41100 MODENA//ITALY/
Journal: COGNITIVE BRAIN RESEARCH, 1993, V1, N2 (APR), P87-93
ISSN: 0926-6410
Language: ENGLISH   Document Type: ARTICLE   (Abstract Available)

...Abstract: and 15 elderly subjects (EC). For the short-time durations we asked the subject to **repeatedly** reproduce a standard interval of 1 s. To test how subjects evaluated longer time durations...

...of verbal estimates of longer durations was severely impaired. AD patients showed increased variability on **repeated** reproduction of 1-s intervals and were both inaccurate and imprecise in their verbal estimate...

...framework of the Scalar Timing Model, we conclude that amnesic patients exhibit a deficit in **encoding** and storing the current time for intervals that exceed their short-term memory range, while...
...Research Fronts: CHOLINERGIC AGENT (HP-128))
92-2579 001 (COCHLEAR IMPLANT; AUDITORY FILTER BANDWIDTHS; TEMPORAL

CUES FOR **CONSONANT** RECOGNITION; MUSICAL TIME; **SOUND** LOCALIZATION; FREQUENCY DISCRIMINATION)
92-3686 001 (IMPLICIT MEMORY; EQUIVALENCE CLASS FORMATION; BEHAVIORAL MECHANISMS IN EVOLUTIONARY...

18/3,K/10 (Item 1 from file: 35)
DIALOG(R)File 35:Dissertation Abs Online

01500101 ORDER NO: AAD96-25190
EVIDENCE FOR THE REPRESENTATION OF SYLLABLES AND SYLLABLE STRUCTURE IN THE PRODUCTION OF NORMAL SPEECH **(PHONOLOGICAL** ENCODING **)**
Author: SEVALD, CHRISTINE ANN
Degree: PH.D.
Year: 1996
Corporate Source/Institution: UNIVERSITY OF ILLINOIS AT URBANA-CHAMPAIGN (0090)
Source: VOLUME 57/04-B OF DISSERTATION ABSTRACTS INTERNATIONAL.
PAGE 2916. 75 PAGES

EVIDENCE FOR THE REPRESENTATION OF SYLLABLES AND SYLLABLE STRUCTURE IN THE **PRODUCTION OF NORMAL** SPEECH **(PHONOLOGICAL** ENCODING **)**

Theories of **speech** production embrace a number of views of the syllable. Some theories do not use syllables...

...whether syllables are chunks that specify their phonological content or schemas that specify an abstract **consonant** -vowel (CV) frame apart from other phonological content. A third view uses both schemas and chunks and is called a mixed view. In four experiments, speakers **repeated** a pair of words or pronounceable nonwords as often as possible in 4-s. Each pair consisted of a monosyllable followed by a disyllable. Subjects' **speech** rate was faster when both the frame and the content of the monosyllable were **repeated** in the first syllable of the disyllable, relative to a condition in which the syllables' sounds were **repeated** but the frame was not. There was no additional advantage for **repeating** both content and structure over **repeating** structure alone. The results support the view that there is an abstract syllable frame that...

...two additional views about syllable schemas. According to the CV-tier view, the vowel or **consonant** status of phonemes is the only featural information represented in the syllable frame. According to...

...vowel occupies two slots, for instance. Part of the new design tested for effects of **repeating** phonological quantity without **repeating** CV structure while the rest replicated the non phoneme-sharing conditions of Experiments 2 and 3. As before, there was a benefit for **repeating** the abstract CV structure. There was also a small benefit for **repeating** only phonological quantity. The results lend support to both views of the schema, with the...

18/3,K/11 (Item 2 from file: 35)
DIALOG(R)File 35:Dissertation Abs Online

748991 ORDER NO: NOT AVAILABLE FROM UNIVERSITY MICROFILMS INT'L.
PHONOLOGICAL AND LEXICAL ENCODING **IN** SPEECH **PRODUCTION: AN ANALYSIS OF NATURALLY OCCURRING AND EXPERIMENTALLY ELICITED** SPEECH **ERRORS**

Author: DELL, GARY SANTMYERS
Degree: PH.D.
Year: 1980
Corporate Source/Institution: UNIVERSITY OF TORONTO (CANADA) (0779)
Source: VOLUME 42/01-B OF DISSERTATION ABSTRACTS INTERNATIONAL.
PAGE 353.

PHONOLOGICAL AND LEXICAL ENCODING **IN** SPEECH **PRODUCTION: AN ANALYSIS OF NATURALLY OCCURRING AND EXPERIMENTALLY ELICITED** SPEECH **ERRORS**

Speech errors, or slips of the tongue, provide important data for an understanding of how **speech** production works. Two classes of errors were studied: **sound** errors, which involve the substitution, deletion, addition, and movement of individual phonemes, clusters, and phonemic...

...aspects of articulation play no causal role in slips.

Three recently proposed models of phonological **encoding** are discussed and tested with respect to their ability to handle **speech** error data. In one analysis of 363 initial **consonant** slips from the Toronto corpus, it was found that the target and substituting **consonants** tended to be similar, particularly with respect to manner of articulation and voicing features. Furthermore, the dimension of similarity of the **consonants** did not differ for anticipatory and perseveratory errors. Another aspect of similarity which was investigated...

...the same 363 errors it was found that the phonemes adjacent to the slipping initial **consonants** tended to be identical, e.g. left hemisphere (--->) heft lemisphere, in which the phoneme /e/ is adjacent to the slipping phonemes, /l/ and /h/. This effect, termed the **repeated** phoneme effect, was experimentally established in the second experiment. This experiment involved the creation of artificial initial **consonant** errors between two words (e.g. mad back (--->) bad mack). Both anticipatory and perseveratory slips were more likely if the vowels following the initial **consonants** were identical (as in mad back) than if they were different (as in mad bake).

An important issue in phonological **encoding** is whether or not **sound** errors tend to create meaningful words, (e.g. barn door (--->) darn bore). The proportion of...

...by chance. This finding, the lexical bias effect, calls into question most current models of **speech** production, which assume that phonological **encoding** is independent of the mental lexicon.

A model of phonological **encoding** is proposed to explain the lexical bias effect. Words and their sounds and meanings are...

...word node connected to the phoneme nodes that spell out that word's phonemes. Phonological **encoding** involves three stages. (1) The activation of word nodes representing words to be said, (2...

...that the model accounts for the basic types of error, the lexical bias effect, the **repeated** phoneme effect, and the effects of changes in the speaking rate. In a test of one of the model's predictions, it was found, as expected, that the **repeated** phoneme effect is true for non-adjacent as well as adjacent phonemes.

Extensions to the...

...of information in the brain. In addition it is suggested that the ultimate causes of **speech** errors are the existence of internal noise and the need for the speech production system to be capable of producing novel **sound** and word combinations.

...

18/3,K/12 **(Item 1 from file: 144)**
DIALOG(R)File 144:Pascal

15090778 PASCAL No.: 01-0250481
Consistency of the auditory nerve response to normal and whispered speech
JUSTIN Stephanie; WICKESBERG Robert
Dept. of Psych., Univ. of Illinois, 603 E. Daniel St., Champaign, IL 61820
Journal: The Journal of the Acoustical Society of America, 2001-05-01, 109 (5) p. 2374
Language: English

Consistency of the auditory nerve response to normal and whispered speech
Multiple presentations of an acoustic stimulus are often used to study the **encoding** of sounds that in the natural environment are not **repeated** . This study examined how consistent the response of the auditory nerve is with respect to...

... trials. Correlation coefficients for the normal syllable were 0.87 and 0.72 during the **consonant** and vowel, respectively. In the whispered condition, correlation coefficients were 0.95 during the **consonant** , but only 0.44 during the vowel. Similar correlations were obtained with comparisons of GAPSTs...

... trials. Correlations obtained from the GAPSTs were higher than those of individual responses for both **speech** conditions and at all intensities. Individual auditory nerve fiber correlations during the **speech** stimuli were variable. These results demonstrate that the use of many repetitions to achieve consistency...
?

21/3,K/1 (Item 1 from file: 6)
DIALOG(R)File 6:NTIS

0105746 NTIS Accession Number: AD-639 964/XAB
Synthetic Speech **Study**
(Final rept. 1 Jul 65-30 Jun 66)
Bogusz, J. ; Smith, R. ; Strohmeyer, G.
Philco Corp Blue Bell P
Corp. Source Codes: 281050
Report No.: 2659-F; ECOM-0300-E-F
30 Jun 66 2p
Journal Announcement: USGRDR6622
See also AD-611 082.
Order this product from NTIS by: phone at 1-800-553-NTIS (U.S. customers); (703)605-6000 (other countries); fax at (703)321-8547; and email at orders@ntis.fedworld.gov. NTIS is located at 5285 Port Royal Road, Springfield, VA, 22161, USA.
NTIS Prices: PC A02/MF A01

Synthetic Speech **Study**
The report describes a technique to automatically evaluate the intelligibility of **speech** transmitted over a communication channel. The technique is called CORODIM (Correlation Of the Recognition Of Degradation with Intelligibility Measurements). It transmits a test signal composed of **speech** -like sounds representative of phoneme **consonants** , and measures, by means of spectral channel analysis, the degradation suffered by each of the...

... signal constituents. The degradation manifests itself as an 'effective noise spectrum' which is measured and **matched** to one of a library of reference noise spectra. By means of the spectrum **matching** operation and a measurement of **signal** -to- **noise** **ratio** each constituent **sound** of the test signal is assigned a probability of recognition. These values are weighted by...

... normalized to obtain a score representative of word intelligibility based on either initial or final **consonant** recognition of CVC-type words. CORODIM evaluates scores for both initial and final **consonants** and takes their product for the overall word intelligibility score. The part of the CORODIM...

... analysis operation was computer simulated. The technique was checked against word articulation scores under identical **speech** link conditions. Two types of **speech** degradations were considered. The first was additive noise having the **same** spectral characteristics as that used in deriving the library of phoneme recognition probability data. The second type was degradation produced under laboratory simulated **speech** link conditions. (Author)
Descriptors: **Speech** representation; * **Voice** communication systems; Effectiveness; Intelligibility; Measurement; Phonetics; Degradation; **Signal** -to- **noise** **ratio** ; **Sound** signals; **Speech** recognition; Probability; Simulation; Programming(Computers); **Speech**
Identifiers: CORODIM; Phonemes; Articulation index; SCIM; **Consonants** ; Words

21/3,K/2 (Item 1 from file: 144)
DIALOG(R)File 144:Pascal

12341712 PASCAL No.: 95-0582917

Villchur revisited: Another look at automatic gain control simulation of recruiting hearing loss

DUCHNOWSKI Paul; ZUREK Patrick M

Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, Massachusetts 02139

Journal: Journal of the Acoustical Society of America, 1995-12, 98 (6) 3170-3181

Language: English

An algorithm to simulate the effects of sensorineural hearing impairment on **speech** reception was investigated. Like that described by Villchur (J. Acoust. Soc. Am. 62, 665-674...

... in the band. In a preliminary evaluation, two normal-hearing subjects listened to the simulation **matched** to hearing losses studied previously (Zurek and Delhorne, J. Acoust. Soc. Am. 82, 1548-1559...

...more-detailed evaluation, the performance of three listeners with severe sensorineural hearing loss on several **speech** intelligibility tests was compared to that of normal-hearing subjects listening to the output of the simulation.These tests included **consonant** -vowel syllable identification and sentence keyword identification for several combinations of **speech** -to-noise ratio, frequency-gain characteristic, and overall level. Generally, the simulation algorithm reproduced **speech** intelligibility well, though there was a clear trend for the simulation to result in better intelligibility than observed for impaired listeners when high-frequency emphasis placed more of the **speech** spectrum above threshold at higher frequencies. Also, the hearing-impaired listener with the greatest loss showed the largest discrepancies with the simulation. Overall, however, the simulation provides a very good **approximation** to **speech** reception by hearing-impaired listeners. The results of this study, toget

English Descriptors: Theoretical study; Algorithms; Auditory organs; **Signal** -to- **noise** **ratio** ; Simulation

Broad Descriptors: Hearing impairment; Signal processing; **Speech** recognition; Trouble audition; Traitement signal; Reconnaissance parole

?

30/3,K/1 (Item 1 from file: 2)
DIALOG(R)File 2:INSPEC

7253355 INSPEC Abstract Number: B2002-06-6130C-003
Title: Joint source-channel decoding approaches applied in adaptive multirate (AMR) speech **transmission**
Author(s): Wen Xu; Marke, M.
Author Affiliation: Dept. of Mobile Phone Dev., Siemens AG, Munich, Germany
Conference Title: IEEE 54th Vehicular Technology Conference. VTC Fall 2001. Proceedings (Cat. No.01CH37211) Part vol.4 p.2514-18 vol.4
Publisher: IEEE, Piscataway, NJ, USA
Publication Date: 2001 Country of Publication: USA 4 vol.(lxxiii+xii+2777) pp.
ISBN: 0 7803 7005 8 Material Identity Number: XX-2001-02310
U.S. Copyright Clearance Center Code: 0-7803-7005-8/01/$10.00
Conference Title: IEEE 54th Vehicular Technology Conference. VTC Fall 2001. Proceedings
Conference Date: 7-11 Oct. 2001 Conference Location: Atlantic City, NJ, USA
Language: English
Subfile: B

Title: Joint source-channel decoding approaches applied in adaptive multirate (AMR) speech **transmission**
Abstract: Investigations are reported on joint source-channel (JSC) decoding applied in the adaptive multirate (AMR) **speech** codec for typical real **speech** . The studied approaches include the "HUK" method, two-step decoding, **repeated** decoding, and **error** concealment based on the so-called soft bit decoding. A method based on a Kalman filter for a priori information estimation is described, where both inter- and intra- **frame** correlation are taken into account. It is shown that for the AMR codec, although highly optimized in exploiting signal redundancy, some redundancy still remains in the **encoded speech** bits. By using an optimized JSC decoding algorithm like the **repeated** decoding, a gain of up to 0.2 dB in channel **signal** -to- **noise ratio** is achievable, compared to decoding without exploiting the residual redundancy. Under bad channel conditions, the **error** concealment algorithm can deliver an even better subjective **speech** quality.
...Descriptors: **speech** codecs...

... **speech** coding
...Identifiers: adaptive multirate **speech** transmission...

... **speech** codec...
... **repeated** decoding...

... **error** concealment...

...channel **signal** -to- **noise ratio** ;

30/3,K/2 (Item 1 from file: 8)
DIALOG(R)File 8:Ei Compendex(R)

05971156 E.I. No: EIP01536789257
Title: Joint source-channel decoding approaches applied in adaptive multirate (AMR) speech **transmission**

Author: Xu, W.; Marke, M.
Corporate Source: Dept. of Mobile Phone Development Siemens AG, D-81675 Munich, Germany
Conference Title: IEEE 54th Vehicular Technology Conference (VTC FALL 2001)
Conference Location: Atlantic City, NJ, United States Conference Date: 20011007-20011011
E.I. Conference No.: 58838
Source: IEEE Vehicular Technology Conference v 4 n 54ND 2001. p 2514-2518 (IEEE cat n 01CH37211)
Publication Year: 2001
CODEN: IVTCDZ ISSN: 0740-0551
Language: English

Title: Joint source-channel decoding approaches applied in adaptive multirate (AMR) speech **transmission**
Abstract: Investigations on joint source-channel (JSC) decoding applied in the Adaptive Multirate (AMR) **speech** codec for typical real speeches are reported. The studied approaches include the "HUK" method, the two-step step decoding, the **repeated** decoding, and the **error** concealment based on the so-called soft bit decoding. A method based on Kalman filter for a priori information estimation is described, where both inter- and intra- **frame** correlation are taken into account. It is shown that for the AMR codec, although highly optimized in exploiting the signal redundancy, some redundancy still remains in the **encoded speech** bits. By using an optimized JSC decoding algorithm like the **repeated** decoding, a gain up to 0.2 dB in channel **signal** -to- **noise ratio** is achievable, compared to decoding without exploiting the residual redundancy. Under bad channel conditions, the **error** concealment algorithm can deliver an even better subjective **speech** quality. 13 Refs.
Descriptors: **Speech** transmission; Communication channels (information theory); **Speech** coding; Decoding; Kalman filtering; Optimization; Algorithms; **Signal** to **noise ratio**
Identifiers: Joint source channel decoding; Adaptive multirate **speech** transmission; **Speech** quality

30/3,K/3 (Item 2 from file: 8)
DIALOG(R)File 8:Ei Compendex(R)

05336894 E.I. No: EIP99084752707
Title: Comparison of iterative **decoder performance with union bounds for short** frame **turbo codes**
Author: Burr, Alister G.; White, George P.
Corporate Source: Univ of York, York, UK
Source: Annales des Telecommunications/Annals of Telecommunications v 54 n 3 1999. p 201-207
Publication Year: 1999
CODEN: ANTEAU ISSN: 0003-4347
Language: English

Title: Comparison of iterative **decoder performance with union bounds for short** frame **turbo codes**
Abstract: We consider short **frame** turbo codes, suitable for delay-sensitive services such as conversational **speech** , or for **encoding** single ATM cells. We compare the uniform interleaver bound of Benedetto and Montorsi, and a...

...weight distribution due to the actual pseudo-random interleaver, with

the simulated BER using an **iterative** decoder. We show that the uniform interleaver bound is significantly pessimistic at high SNR, where...
Descriptors: Codes (symbols); **Error** correction; Decoding; Asynchronous transfer mode; **Speech** communication; **Iterative** methods; Bit **error** rate; Computer simulation; **Signal** to **noise** **ratio**

30/3,K/4 (Item 3 from file: 8)
DIALOG(R)File 8:Ei Compendex(R)

05235688 E.I. No: EIP99020017537
Title: Optimal coding rate of punctured convolutional codes in multiservice wireless cellular systems
Author: Gauvreau, Jean-Louis; Despins, Charles L.; Yang, Jun; Delisle, Gilles Y.
Corporate Source: Microcell Connexions, Inc, Montreal, Que, Can
Source: IEEE Transactions on Vehicular Technology v 48 n 1 Jan 1999. p 115-125
Publication Year: 1999
CODEN: ITVTAB ISSN: 0018-9545
Language: English

Abstract: The microcellular link performance of future multimedia wireless systems could be improved by using **error** -correcting punctured convolutional codes in conjunction with slow-frequency hopping. However, the bandwidth expansion due...

...for a given bandwidth allocation. This work determines the best compromise between the power of **error** correction due to coding and the strength of the self-induced system interference in terms of numerous criteria for **speech** and data transmission. The aforementioned tradeoff is evaluated in terms of the average bit **error** rate (BER), the **frame** **error** rate, and the burst **error** distribution for **voice** transmission. For data transmission with a type 1 hybrid selective- **repeat** automatic **repeat** -request (ARQ) protocol, the criteria are average throughput and throughput distribution, the round-trip acknowledgment...
Descriptors: Convolutional codes; **Encoding** (symbols); Cellular radio systems; Wireless telecommunication systems; Frequency hopping; Bandwidth; **Signal** to **noise** **ratio** ; Frequency division multiple access; Time division multiple access; Radio links
Identifiers: Punctured convolutional codes; Automatic **repeat** -request (ARQ) protocol

30/3,K/5 (Item 1 from file: 35)
DIALOG(R)File 35:Dissertation Abs Online

01515770 ORDER NO: NOT AVAILABLE FROM UNIVERSITY MICROFILMS INT'L.
AUTOREGRESSIVE MODELLING FOR SPEECH **CODING: ESTIMATION,** INTERPOLATION **AND QUANTISATION**
Author: ERKELENS, JOHAN STEFAN
Degree: DR.
Year: 1996
Corporate Source/Institution: TECHNISCHE UNIVERSITEIT TE DELFT (THE NETHERLANDS) (0951)
Source: VOLUME 57/04-C OF DISSERTATION ABSTRACTS INTERNATIONAL. PAGE 1379. 144 PAGES
ISBN: 90-407-1338-3

Publisher: DELFT UNIVERSITY PRESS, STEVINWEG 1, 2628 CN DELFT, THE NETHERLANDS

AUTOREGRESSIVE MODELLING FOR SPEECH **CODING: ESTIMATION,** INTERPOLATION **AND QUANTISATION**

This thesis focuses on different aspects of autoregressive modelling for **speech** coding, mainly estimation, **interpolation** and quantization.

Different autoregressive estimation methods are discussed and it is shown that the well...

...the variance of the models. Other methods are available that do not need a window.

Interpolation of the LPC model is investigated. It is shown that certain representations of the model are not suitable for **interpolation** . Moreover, a new **interpolation** method is developed that incorporates the energy of **frames** in the **interpolation** procedure.

The distortion measures that are commonly used for quantization of the parameters of the...

...commonly used distortion measures belong to a class of similar distortion measures for small quantization **errors** . The results explain why the Line Spectrum Frequencies are more suited for quantization than other...

...parametric LPC distortion measures.

It is investigated if the CELP algorithm can be improved by **iteratively** adapting the LPC model and the excitations to each other. An increase in **Signal** to **Noise** **Ratio** was found, but no increase in subjective quality, because the LPC model loses its interpretation in terms of an accurate description of the **speech** spectral envelope.

A new distortion measure is developed that does not belong to the class...

?

36/3,K/1   (Item 1 from file: 35)
DIALOG(R)File 35:Dissertation Abs Online

01889924 ORDER NO: AADAA-I3053571
Toward a psychologically and computationally adequate model of speech **perception**
  Author: Webster, Gabriel Jesse
  Degree: Ph.D.
  Year: 2002
  Corporate Source/Institution: University of Washington (0250)
  Source: VOLUME 63/05-A OF DISSERTATION ABSTRACTS INTERNATIONAL.
           PAGE 1816. 139 PAGES
  ISBN: 0-493-68411-5

Toward a psychologically and computationally adequate model of speech **perception**

...information, as revealed by experimental data. Computational adequacy reflects a concern that models use real **speech** input, that they be explicit, and that they produce reasonable output. I focus on bridging ...

...on the one hand and computationally adequate on the other by developing a theory of **speech** perception, called <italic> Alpaca</italic>, that explicitly assumes that the input is noisy and variable. The mechanism that gives Alpaca the crucial ability to deal with noisy and variable **speech** is the use of a measure of <italic>confidence</italic>, which is a subconscious measure of how confident the **speech** processor is that the **encoding** of some input signal as some output phone is correct. Confidence reflects two dimensions of the input signal. The **first** dimension I call <italic> certainty</italic>, and depends on how reliably a given cue is extracted from the signal. In noisy conditions, cue extractability, and thus certainty, decreases. The **second** dimension is <italic>goodness of fit</italic>, and refers to how **close** the **match** is between an input and a stored target. Goodness of fit relies on **matching** input stimuli to <italic>dynamic targets</italic> that constantly move to locations that are appropriate...

...solution to the problems of noise and variability in the signal, thus supplying a theoretical **framework** to guide and situate implementation-level research.

36/3,K/2   (Item 2 from file: 35)
DIALOG(R)File 35:Dissertation Abs Online

01367405 ORDER NO: AAD94-22376
COMPUTATIONAL MODELS OF THE PROSODY/SYNTAX MAPPING FOR SPOKEN LANGUAGE SYSTEMS
  Author: VEILLEUX, NANETTE MARIE
  Degree: PH.D.
  Year: 1994
  Corporate Source/Institution: BOSTON UNIVERSITY (0017)
  Source: VOLUME 55/03-B OF DISSERTATION ABSTRACTS INTERNATIONAL.
           PAGE 1106. 238 PAGES

Prosodic information, **encoded** in **speech** as the grouping of words (phrasing) and the relative prominence of some syllables in an utterance,

is important in human understanding of **speech** . In order to use prosodic information in automatic spoken language systems, computational models of the...

...between the acoustic and syntactic domains. The joint acoustic/prosody/syntax model is used in **speech** understanding to compute a prosody-parse score, which expresses the degree of the **match** between acoustic features and a proposed syntactic representation.

One major contribution of this work is that the computational models are formulated in a probabilistic **framework** that uses decision trees in a non-traditional way, to estimate probability distributions. The models themselves represent a significant contribution in part because each demonstrates that the **same** models can be used in both synthesis and understanding applications. The usefulness of these models is demonstrated in three applications. **First** , the decision tree and the hierarchical model are used to predict the correct placement of prosodic phrase boundaries, exploiting the relationship between prosody and syntax to improve synthetic **speech** quality. **Second** , the probabilistic prosody-parse scoring system is used to automatically select between two possible interpretations of an utterance, achieving performance **close** to that of human listeners. Finally, the prosody-parse scoring system is used in an existing automatic **speech** understanding system to improve word recognition performance. Although their utility is demonstrated in specific implementations...

36/3,K/3 (Item 1 from file: 144)
DIALOG(R)File 144:Pascal

14757148 PASCAL No.: 00-0435062
The processing of derived and inflected suffixed words during reading
Cross-linguistic perspectives on morphological processing
NISWANDER Elizabeth; POLLATSEK Alexander; RAYNER Keith
FROST Ram, ed; GRAINGER Jonathan, ed
University of Massachusetts, Amherst, United States
Department of Psychology, The Hebrew University of Jerusalem, Jerusalem, Israel; CNRS and University of Provence, Aix-en-Provence, France
Journal: Language and cognitive processes, 2000, 15 (4-5) 389-420
Language: English

The **encoding** of suffixed words (both derivations and inflections) was assessed by monitoring eye movements during reading...

... and root frequency were independently manipulated, where pairs of words differing on one variable and **matched** on the other were inserted into the **same** sentence **frame** . For derived words, root morpheme frequency affected processing earlier than did whole-word frequency: it affected the duration of the **first** fixation, whereas whole-word frequency affected processing only beginning on the **second** fixation. In contrast, for (the regular) inflected words, word frequency had significant effects beginning on the **first** fixation, whereas root frequency had significant effects beginning with the **first** fixation duration only for plural nouns and not for inflected verbs. **Subsequent** regression analyses on the inflected words suggested that the usual part of **speech** for the stem may play a significant role in processing. The data thus indicate that...
?

40/3,K/1 (Item 1 from file: 2)
DIALOG(R)File 2:INSPEC

03832923 INSPEC Abstract Number: B91018757
Title: Combined source and channel coding based on multimode coding
Author(s): Taniguchi, T.; **Amano, F.** ; Unagami, S.
Author Affiliation: Fujitsu Lab. Ltd., Kawasaki, Japan
Conference Title: ICASSP 90. 1990 International Conference on Acoustics, Speech and Signal Processing (Cat. No.90CH2847-2) p.477-80 vol.1
Publisher: IEEE, New York, NY, USA
Publication Date: 1990 Country of Publication: USA 5 vol. 2970 pp.
U.S. Copyright Clearance Center Code: CH2847-2/90/0000-0477$01.00
Conference Sponsor: IEEE
Conference Date: 3-6 April 1990 Conference Location: Albuquerque, NM, USA
Language: English
Subfile: B

Author(s): Taniguchi, T.; **Amano, F.** ; Unagami, S.
...Abstract: to source and channel coding, is presented. The optimum coding mode is selected in each **frame** , based on an evaluation of the spectral distortion (SN/sub LAR/) in reproduced **speech** . The threshold value of SN/sub LAR/ for mode decision is varied according to the...
...Descriptors: **speech** analysis and processing
...Identifiers: **speech** coding

40/3,K/2 (Item 1 from file: 94)
DIALOG(R)File 94:JICST-EPlus

00294282 JICST ACCESSION NUMBER: 86A0436056 FILE SEGMENT: JICST-E
A study of variable rate coding with ADPCM-MQ.
TANIGUCHI TOMOHIKO (1); UNAGAMI SHIGEYUKI (1); **AMANO FUMIO** (1); OKAZAKI KOJI (1)
(1) Fujitsu Labs. Ltd.
Denshi Tsushin Gakkai Gijutsu Kenkyu Hokoku, 1986, VOL.86,NO.77, PAGE.57-64(SP86-20), FIG.9, TBL.3, REF.4
JOURNAL NUMBER: S0532BAP
UNIVERSAL DECIMAL CLASSIFICATION: 621.391.037.3 681.3:801.4
LANGUAGE: Japanese COUNTRY OF PUBLICATION: Japan
DOCUMENT TYPE: Journal
ARTICLE TYPE: Original paper
MEDIA TYPE: Printed Publication

TANIGUCHI TOMOHIKO (1); UNAGAMI SHIGEYUKI (1); **AMANO FUMIO** (1); OKAZAKI KOJI (1)
ABSTRACT: In this paper, we propose a variable rate **speech** coding algorithm which varies the coding bit rate **frame** by **frame** to obtain a high coding efficiency. The proposed algorithm has several ADPCM coding blocks with...
...of each locally decoded signal, the optimum coding block is selected and switched in each **frame** . We studied about two measurement of evaluation, that is, segmental S/N and noise level...
...DESCRIPTORS: signal **frame** ; ...

... **speech** compression
...BROADER DESCRIPTORS: **speech** processing
?